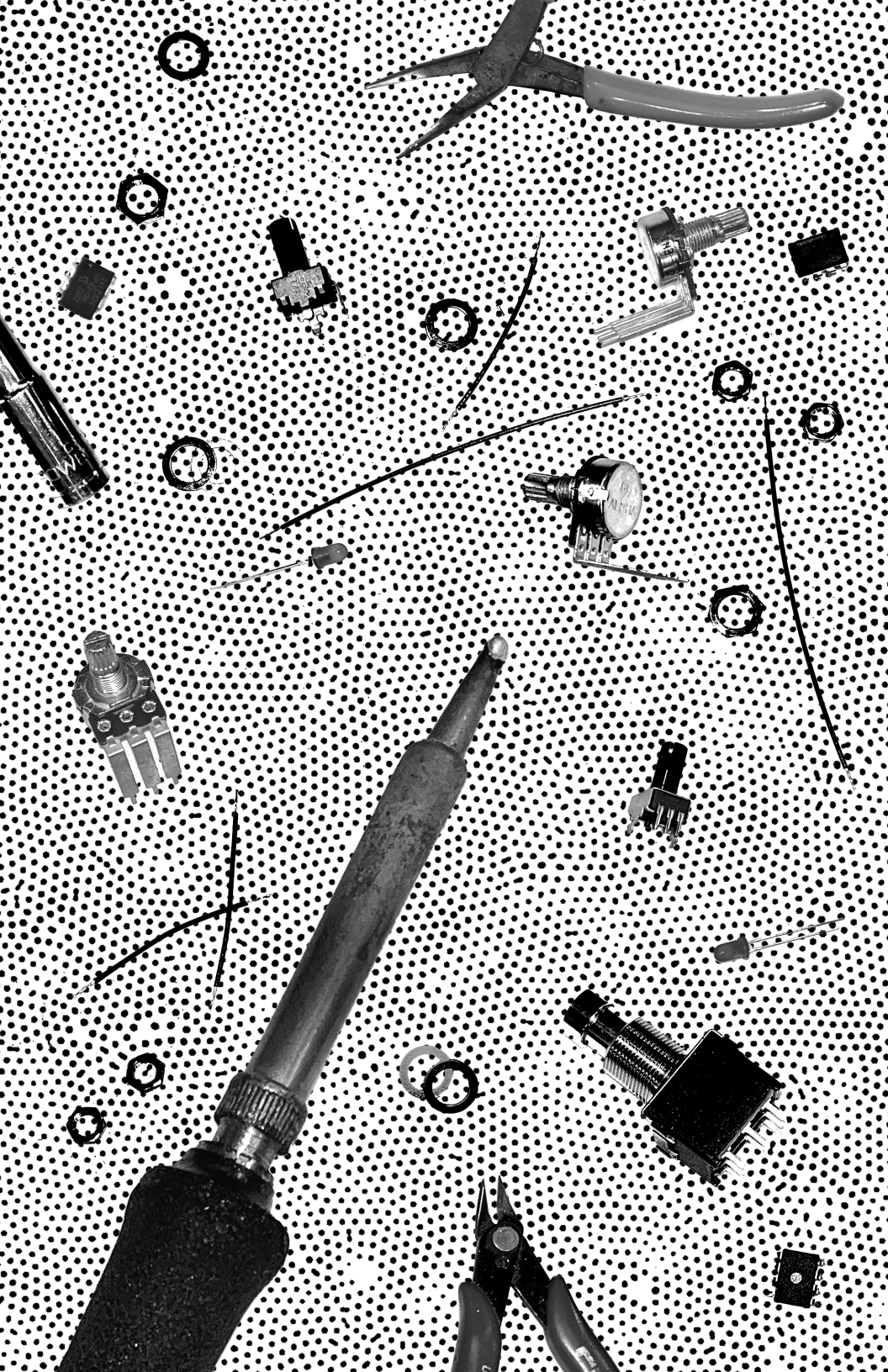
circuit punk

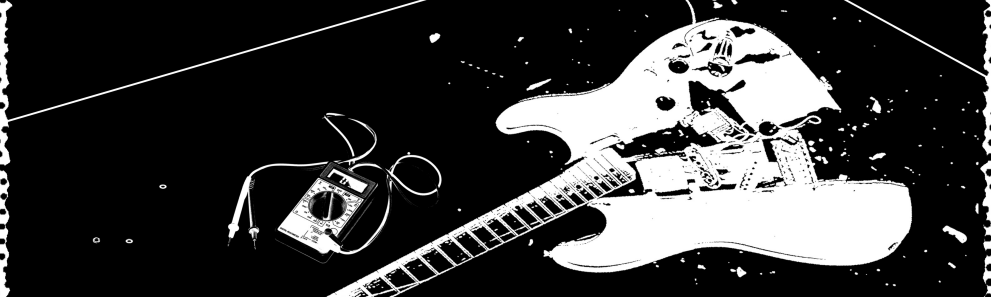
A MUSIC TECHNOLOGY ZINE

ISSUE #001



CONTENTS

Unusual Comparator FX	1
Augmenting The Guitar	3
Mute It.	7
LTC1799 Mods	11
Adding Artwork To PCBs	15
Raging Droner	17
Contact: A Simple Chaotic Microphone	19
Interview: Psychiceyeclix	22
DRUGStm's Circuit Bent PS2	26
Multiband Saturator	32
Reverse Active Transistors	35
Getting Loopy With Tape Loops	41
3D Print Guitars	43



UNUSUAL COMPARATOR FX

by Cameron Glass
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Comparators are highly underrated. Here I'll demonstrate a couple of their hidden gems. First, a review of the basic function and effect of a comparator.

The Basics

By configuring an op amp with no feedback, the op amp acts as a comparator. Comparators drive the output high when the non-inverting input is larger than the inverting input, and drive the output low in the opposite case. This makes them perfect for hard clipping distortions, like that shown in figure 1a. Just put some audio signal into one input, and a DC voltage somewhere around its midpoint into the other input, and voila: It's squashed into a square wave. For guitar effects, a boost before the comparator allows it to sustain oscillations for longer periods of time (fig. 1b).

A nice thing about this type of distortion is that its output is gated when the input signal is sufficiently quiet, such as when a guitar player mutes their strings. And, as the guitar signal dies out after striking a chord, it chops into silence in a glitchy way.

It should also be noted that your dynamic range gets squashed to 0- It's either all the way loud, or completely silent.

Pulse Amplitude Modulator

All that's needed to turn a basic comparator distortion into something like a pulse amplitude modulator is another audio source like an oscillator, and (optionally) an additional comparator. Take the guitar pedal circuit shown in figure 2 for example. In this example, I use a square wave oscillator running on a single-supply 5V. If you want to build this circuit, practically any oscillator will do. I recommend a simple 555 or CD4046 based one, since those can easily run on a single supply.

We re-bias the basic comparator output so it's in the middle of the final comparator's supply range. Its amplitude doesn't matter too much, but it's about 3.5V peak to peak in this case. The oscillator signal at the other input is attenuated down to the 0-2.4V range. The important thing here is that it peaks slightly below the centerpoint of the guitar signal (2.5V in this case). This makes it so when the guitar's strings are muted, the two signals never overlap, which gates the output like a true amp mod. When the guitar does play, its signal rapidly overlaps the oscillator, making the desired effect at the output.

The comparator between the boosted guitar signal and the final output comparator is only necessary for this particular effect. Removing it, you get a more tonally dynamic effect with a shorter sustain. If you try this, you may want to attenuate the oscillator signal even closer to 2.5V for a longer sustain.

Harmonic Sweep

There's an interesting way to get a comparator distortion to morph into a fifth-up harmonic. First, consider that by adjusting the voltage at the inverting input of the comparator in figure 1b, you can effectively narrow one half cycle of the square wave output.

The circuit shown in figure 3 builds on this idea by replacing that input with an additional comparator. By having this signal attenuated towards (a heavily filtered) 2.5V, you can sweep both the positive and negative half cycles of the audio. I'll leave it to you to see what this sounds like.

Conclusion

I likely didn't invent these circuits, but I didn't get them from anywhere either. There's evidently tons of ways to combine, modify, and destroy your signals using comparators alone. Hopefully this inspires you to come up with your own weird shit!

fig. 1a

generic comparator distortion
(dual supply)

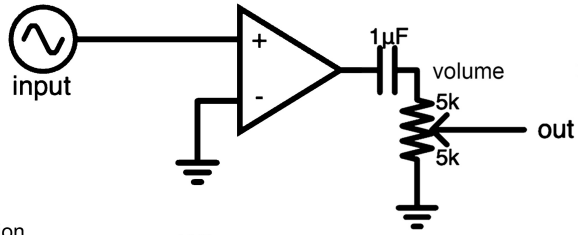


fig. 1b

basic comparator distortion
for guitar

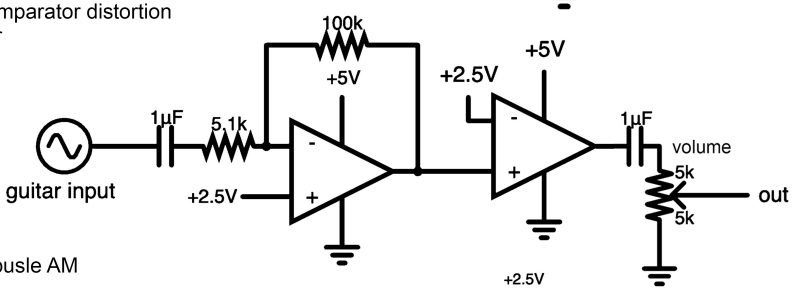


fig. 2

pseudo pusle AM
for guitar

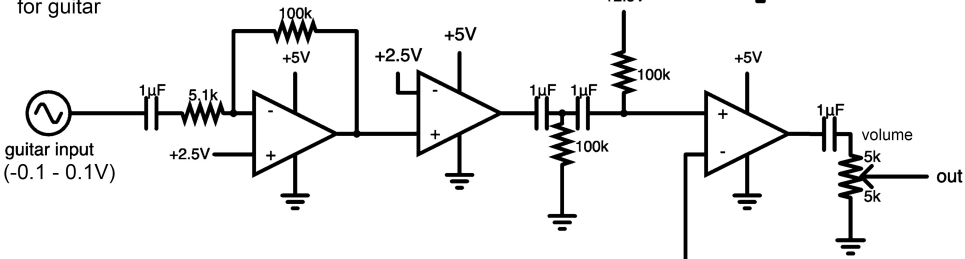
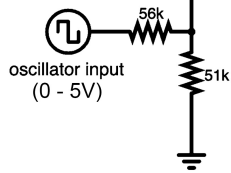
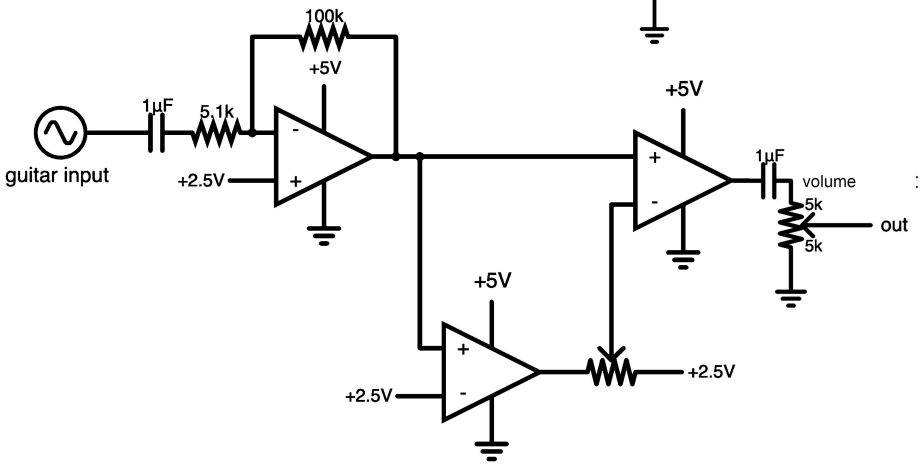


fig. 3

harmonic sweep
for guitar





AUGMENTING THE GUITAR

PHYSICAL TRANSFORMATIONS TO ENABLE SONIC TRANSFORMATIONS

James Corley

For a long time, I have been driven to explore the limits of the guitar's sonic potential. Through digital processing and physical gestures, I have wondered whether the guitar's capabilities could be expanded without losing the idiosyncrasies that make it a recognisable and much-admired instrument. This curiosity led to my PhD research at Keele University (Staffordshire, UK), where I am investigating how the physical and sonic transformation of a guitar can impact compositional and performance practices. In doing so, I ask whether the guitar can embody musics and sounds that are often disconnected from its traditional history—such as experimental electronic music and sound design.

Given the complexity of this multifaceted, cross-disciplinary project, this article focuses on the preliminary technical aspects of realising such transformations.

Planning

The project's primary requirement was clear from the start: achieving multidimensional touch control of sound processing for each string independently. Having per-string processing opens up many possibilities that can be creatively taken advantage of, but poorly built hexaphonic systems would have unacceptable levels of cross-talk (i.e. the 1st string bleeding into the pickup of the 2nd string). Initial prototypes involved modifying a Fender Stratocaster with a CycFi Nu-Multi 6[1] hexaphonic pickup system to test the feasibility of hexaphonic processing. This system is exceptionally well-designed, carrying the signals of all six strings through a single 19-pin cable, which also powers the active pickup system via the CycFi Nexus. The Nexus separates these signals into individual 1/4-inch connections, which can then be routed into a multichannel audio interface (a Behringer UMC1820 in this case). For multidimensional control, I selected Bela Trill[2] capacitive touch sensors. These sensors can track various touch parameters—such as X/Y position and touch diameter—for multiple fingers. When connected through an Arduino, these signals can be translated into any required data for sound processing.

Determining the appropriate digital signal processing (DSP) platform proved challenging. Audio programming languages like Supercollider and Pure Data running on a Raspberry Pi lacked sufficient power to handle the complex processing combinations I envisioned. Standalone multi-effects units, such as the Line 6 Helix[3], were unable to manage six independent inputs, and their MIDI mapping capabilities were limited. Similarly, the Symbolic Sounds Kyma[4] system had workflow and design paradigms that felt ill-suited to the project's needs. Ultimately, I determined that a general-purpose computer running Max/MSP[5] within Ableton Live[6] provided the flexibility, power, and integration required to fit this system into modern music production workflows.

[1] https://www.cycfi.com/product/nu-multi-6/

[2] https://bela.io/products/trill/

[3] https://line6.com/helix/helix-rack.html

[4] https://kyma.symbolicsound.com/pacamara-apu/

[5] https://cycling74.com/

[6] https://www.ableton.com/en/live/

Construction

Once the concept proved viable, I began planning the full instrument, ensuring every decision aligned with the research goals. A 27-inch scale length was chosen to allow the guitar to be tuned down to C standard; because this is nearly half way between E standard guitar tuning and E standard bass tuning, the amount of shifting required to reach different tunings within that range is reduced, thus reducing the artefacts inherent when pitch shifting by large amounts. However, this choice introduced unforeseen consequences (see Section 4). The nut width was set to 48mm—unusually wide for an electric guitar but more akin to classical guitar construction. This spacing supports cleaner contrapuntal playing on adjacent strings, a style that benefits from independent string processing. The neck was designed as a bolt- on for easy maintenance.

The body design drew inspiration from traditional hollow body instruments, incorporating a hollow body (with a full upper horn) to accommodate numerous controls and internal electronics without spatial constraints. A red translucent Perspex top houses the electronics, ensuring modularity for future upgrades without requiring woodworking modifications.

Construction was done by Ancoats Guitars[7], a Manchester-based luthier team made up of Dave Roberts and Hugh Ford[8].

The body is poplar, the neck and fretboard are maple, and the top is a translucent perspex. The guitar was fully designed in CAD software by Hugh before being CNC'd. Structural supports had to be added for stability, with the added bonus that the supports were visible through the perspex top, making the guitar look very unique The neck, hand-carved and stained black, features a natural relic appearance due to the satin finish not adhering evenly to the maple. The Bela Trill controls were affixed to the Perspex using strong double-sided tape, but requiring slight sanding for a flush fit.

The instrument was strung with .14 gauge nylon tape-wound strings to create a darker tone that contrasts with the full-range pickups and to reduce the string handling noise that such powerful pickups amplify.

Electronics

The pickup system in the guitar is fully self-contained, and is unmodified from the factory. The Nu-Multi 6 plugs into an internal breakout board which handles all power and audio connections to the 19-pin output. A Bareknuckle P90[9] is in the neck position, also routed into the breakout board as the board can accept extra audio sources, however, due to the lack of shielding in the guitar, the P90 is too susceptible to noise. The Nu-Multi however, is not, which is part of what makes the Nu-Multi system so

Machining the body | Finish stained body | Carved neck | Staining the neck | Sanding the touch pads down

[7] https://www.ancoatsguitars.co.uk/

[8] https://www.instagram.com/hughmakesthings/

[9] https://www.bareknucklepickups.co.uk/pickup/hsp90/supermassive

appealing.

The control system posed significant technical challenges, requiring extensive testing to ensure compatibility, minimise interference, and achieve stable operation. At the heart of the system are seven Bela Trill capacitive touch sensors. They communicate over the I2C bus, using 4-pin JST connectors to link to a central board (the Bela Hub). This hub connects to the SDA (data line) and SCL (clock line) pins of an Arduino Nano, which serves as the primary controller for the guitar's control system.

In addition to the Trill sensors, the system includes other input devices to enhance interactivity:

• Two SHARP infrared proximity sensors[10] are connected to the Arduino's analog pins. These sensors detect the distance of objects, such as the player's hand, allowing for gesture-based control.

• A CycFi 5-way switch[11], also connected to the analog pins, provides quick access to a 5 step parameter control.

• An arcade-style button, wired to a digital pin, offers tactile feedback for momentary parameter changes.

One of the key design considerations was the power and data connection for the Arduino. Several options were explored, including using a traditional 5-pin MIDI socket powered by an internal battery. However, since the guitar was intended to be used exclusively with a computer, USB emerged as the optimal solution. USB not only provides a reliable power source but also facilitates seamless data transfer, allowing the Arduino to send either MIDI or Open Sound Control (OSC) data directly to the computer.

The reliance on USB connectivity does come with trade-offs, as the guitar cannot function as a standalone instrument. It's incompatible with traditional pedalboards, amplifiers, or standard MIDI devices. Instead, it is designed specifically to integrate with a custom software system built in Max/MSP. This system receives control data from the Arduino over USB and processes it into OSC or MIDI signals to manipulate sound in real-time. While this reliance on external software limits the instrument's standalone capabilities, it provides unparalleled flexibility for integrating advanced control techniques into contemporary music production workflows.

Final Thoughts

During testing, some design assumptions proved problematic. For instance, when deciding on a low C standard tuning, I didn't take into account that adaptive granular pitch shifters require at least two cycles of the input frequency to track pitch accurately. This introduces latency, which increases the lower the input frequency. The latency isn't egregious, although a higher tuned guitar would likely yield a better feeling for the player. Another advantage of tuning the guitar higher is the added tension creates a brighter and cleaner signal for the processing to work with.

Additionally, the guitar's ergonomics require significant refinement. A second version would prioritise physical comfort and ease of setup by reducing the body and neck size, streamlining controls, and leveraging the breakout board's compatibility with control voltage sources. Controls should also be repositioned closer to the right hand for more intuitive access. A pressure-sensitive pick, currently under development, could further enhance parameter control without interrupting playing, and using more momentary buttons could provide a simplified and more tactile control system. Addressing these ergonomic/functional issues in a future iteration will bring the instrument closer to being a fully integrated tool for experimental composition and performance.

In terms of connectivity, requiring custom software built in Max/MSP feels unnecessarily restrictive in the long term. Setting the guitar up with direct MIDI out

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10 https://thepihut.com/products/sharp-gp2y0a41skof-infrared-distance-sensor-4-30cm

11 https://www.cycfi.com/product/cv-switch/

(either by USB or 5-pin DIN) would make it more applicable to different environments. Experiments with plugin hosts such as Apple's Mainstage have shown that taking advantage of pre-existing ecosystems is beneficial to the creative process, as the artist can focus on creating, rather than developing.

The final oversight was the modularity of the system - whilst in theory everything is fully modular - the perspex top and electronic systems are actually incredibly difficult to modify without completely dismantling all the electronics, due to how the internal wiring connects between the internal cavities. This is far from ideal, and makes modifications and maintenance of the electronics more stressful than it needs to be.

The Final Build

(To see the guitar in action, go to https://www.instagram.com/redeveredev/ & https://www.youtube.com/@redeveredev)

MUTE IT.

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I've been designing a performance mixer in Eurorack format for a while now. It's been years. Yes, I have issues with keeping focus sometimes and often get distracted by other more shiny things. It's one of those projects that is just taking forever to actually get anywhere as there are many little things that need to be considered. One of those supposedly small things that led me on a wild goose chase for the ideal solution was the humble mute switch. Any mixing desk worth its salt needs a channel mute and solo function if you ask me, and what initially started out as a simple thing quickly evolved into a whole wonderland of things I hadn't even considered at first.

Down the rabbit hole…

Everyone who has dealt with the mixers on one occasion or another knows the occasional 'click' or 'pop' which happens when you mute or unmute an audio signal. These are transients caused by rapid voltage changes in the audio signal when it's interrupted instantaneously. In ELI5 language, the voltage drops from wherever it is to 0 immediately, and that causes an audible transient. Many mixers are prone to this since they use switches, or in rare cases, relays to physically interrupt the signal flow. Those are very effective solutions, as they completely break the signal path. And since they add no active components in the signal chain, they don't introduce any additional distortion, so they're clear as a whistle when it comes to signal integrity. But most importantly, it's the cheapest solution. If you need to implement something for every channel on a mixing desk, component price becomes increasingly important as channel count grows.

In most situations this solution is just fine. No one bats an eye over an occasional plop, unless you're performing in front of an audience. So, it's something to consider on a performance type of mixer. Since I like to use mixing desks actively during live sessions, really bashing that mute button like there is no tomorrow, whatever I design should be able to cope. For that to be possible, though, we actually need a circuit that gradually turns the signal down, quickly enough for our ears to perceive it as instantaneous, but slowly enough to avoid the transient. From what I've read online, around 10 ms would do the trick, fast enough to still be considered instant, yet long enough to have a sort of fade-out, preventing transient noises.

The first idea (and that's seldom a good one) that comes to mind is implementing a VCA. They come in handy packages like the AS3360, V2164, THAT 2180, … or less handy packages like the LM13700. The obvious problem with the VCA approach is that it is not ideal from a cost perspective for everything but the top-tier desks. They also require a substantial amount of extra circuitry and development as you really need them to be as clean as possible since they are squarely in the signal path, and any noise they generate will be added together with all the noise from all the other channels. If you're not careful, you end up with an expensive noise generator.

A VCA is a ton of work and cost that's hardly justifiable for a simple mute switch. So, how can we do it differently?

So what?

If we could use some sort of variable resistor we can turn a simple inverting opamp configuration into something

resembling a VCA. When you say variable resistor, half the internet yells vactrols back at you — Sidenote: A Vactrol is the combination of a photosensitive resistor paired with an led. if the led is on, the resistance is low, if it's off, the resistance is high. — It's what the DivKids Mute does so it seems, but is it actually a good idea in this case, let's look at the idea a bit closer.

Depending on the type of vactrol, they can go from a few Ohms when on to several MegaOhms when off and everything in between. It has some other properties which are useful for this kind of purpose. While the reaction time is often a problem when dealing with vactrols, it's now something that's desirable. We would not need to implement some way of slowing down the transition as those things just cannot transition too quickly even if you wanted them to.

But it will still leak like Grandma with bladder issues. Trying its best, but to no avail. Since the resistance of a vactrol in the off state is not infinite, some audio will still get through and it's more than we want. In order to improve the characteristics, we can put another one, but that one shunts the signal to ground and should be on when the other one is off and vice versa. That way, it would provide a low resistance path to ground when the mute is activated, further lowering the signal off state. This way, we can be pretty sure the remaining signal is now low enough to be buried in the noise floor so deep it'll never see the light of day again.

Seems like a really nice solution to the problem, but as with all things in life, there is always a "but" — not butt you perverts. In this case, the but is rather big.
They're illegal. At least in the EU. They contain Cadmium, which is a hazardous material that is banned in the EU for consumer electronics. Even for non-EU citizens, they will become harder and harder to come by as their usage plummets in the next few years, and they haven't been used much in commercial products anyway for at least a decade or three anyways. But even without the EU restrictions, vactrols are expensive, physically big components and you need to resort to specialized stores to buy them, as they're not available on the big electronic component stores. The VTL5c3 is the most commonly used in audio applications, and last I checked, they're around $2-$3 a piece. If you need 2, that's not any cheaper than a VCA chip.

You can make them yourself if you buy some LDRs, LEDs, and black tape, so it's perfectly fine for DIY projects, but that approach is hardly usable for anything more than a one-off.

So what's next?

On to the next candidate. A JFET can also be used as a crude variable resistor. As with the vactrols, resistance is not infinite in its off state nor 0 in the on state, so we can apply the same trick again of shunting one to ground and placing one in the audio path and get great mute characteristics that way. There will always be something coming through, but if the signal gets low enough, it will just get lost in the noise floor. Using the JFETs in current mode like this allows us to stop worrying about the amplitude of the signal, which would introduce a bit of other things we need to take into consideration, but why make our life more complicated than it should be?

FETs will react a lot faster than vactrols do, so we need to address that issue; otherwise, we have just found a very convoluted way to mute things without gaining much over a simple switch. JFETs do know shades of grey and can be somewhere between on and off, while the characteristics are not great when it comes to linearity and distortion; it's not really an issue since we'll be switching fast enough, and the added distortion during the switch will be unnoticeable. When I say distortion, a lot of people actually think about loud sounds; in the technical sense, distortion is just the signal being changed in a non-linear fashion, which adds some harmonics and has nothing to do with how loud something is.

Figure 1 is the circuit I ended up using (slightly modified for simplicity).
As you can see, the control signal goes through a simple low-pass filter, which introduces a bit of a slope in the signal, which translates into it taking some time for the JFETs to change state. A simple op-amp is used as a comparator to provide an inverted control channel to the 2nd JFET, causing one to have low resistance when the other has high resistance. The diodes are there to prevent the signal going positive, as that would leak DC into the signal path, and we don't want that. There are other ways of generating the control signals, using logic ports for example or generated by an MCU, just be aware that they need to be lower than the signal level in order to turn off the JFET — in practice that means you need a negative voltage — so there will always be a bit of extra trickery to achieve that.

The J series JFETs are designed for signal switching, so it's not a real surprise to see them used just for that purpose. The J111 offers the lowest ON resistance (about 30 Ohms), and that's why I chose it, but it does require at least -10V to be able to turn off completely. In Eurorack land, that's perfectly doable as we have access to -12V easily, but that might be an issue in other designs. The J112 is also being used often in these kinds of applications, having a slightly higher ON resistance, but with a lower voltage requirement to turn off, which might be simpler to implement in some cases.

It's trivial to expand this circuit so it can be used for example as a signal switcher or double it for stereo usage. It's also compact enough, at least with SMD components, to be used throughout a mixing desk in any applications where silent muting/switching is desired/required. In my design I implemented them for the solo feature as well, while I went for simple switches in other parts of the routing features (like post/pre aux send switches).

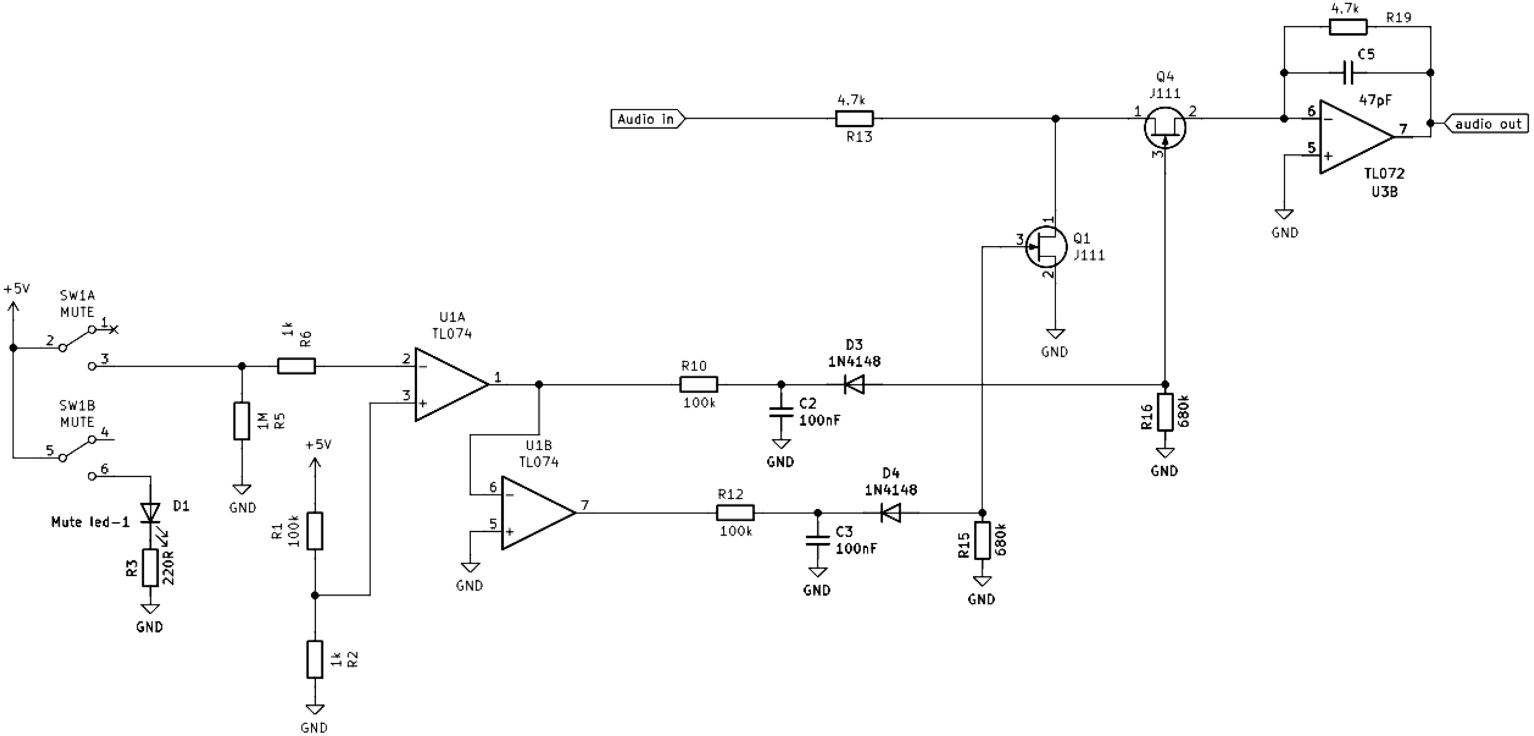
Conclusion

There are still a lot of other ways one could dream up that could be used to mute a signal, all with their own sets of pros and cons. I've restricted myself to only a handful of options, but CMOS gates (like the CD4066) come to mind as another way to do these kinds of things. It's also possible to design a variety of other circuit topologies using JFETs or something else that can serve this purpose. I've tried various approaches and settled on this one, but that doesn't mean there aren't other solutions to the issue which fit certain designs better than this approach. All things considered, the JFET solution seemed an elegant way at a reasonable cost and footprint size. As with everything in electronics, you kind of need to find the sweet spot between quality, functionality, cost, circuit size, and a bunch of other requirements that escape me right now.

Skull & Circuits Bio

Hi, I'm Johan, I'm the designer, owner, manager, logistics operator and janitor of Skull & Circuits, a small Belgian based Eurorack module manufacturer which also dabbles in synth repair and occasionally makes some music. I'm not an engineer by education and thus not formally qualified to write this kind of article, but nevertheless here I am, typing away anyways.

fig 1


Audio in
4.7k
R13
Q4
J111
4.7k
R19
C5
47pF
TL072
U3B
audio out
GND
Q1
J111
GND
+5V
SW1A
MUTE
SW1B
MUTE
Mute led-1
D1
R3
220R
GND
1M
R5
1k
R6
U1A
TL074
U1B
TL074
+5V
R1
100k
1k
R2
GND
R10
100k
C2
100nF
GND
D3
1N4148
R16
680k
GND
R12
100k
C3
100nF
GND
D4
1N4148
R15
680k
GND

LTC1799 MODS

Po8aster
https://petercatproductions.com/

LTC1799 Mods are usually called Clock Mods or Pitch Mods. Basically, you use a fancy chip (the LTC1799) to hijack the oscillator that controls the speed at which a device runs. By underclocking or overclocking a musical device, it will begin warping the sounds produced, usually pitching them up or down as you change the clock speed up or down.

Figure 1: A Clock Modder LTC1799 Board

That might sound super involved, but the beauty of using this particular fancy chip is that it's actually super easy to do and has a ton of different applications.

Let's talk first about what an oscillator does in this context. It's not the synthesizer kind you might think of; instead think of it like a heartbeat or metronome connected to a computer. Or one of those big drums they beat so that the buff dudes all row at the same time; whatever works for you. As the speed of the metronome/drum goes up, the musician plays faster/the rowers row faster; and as the beat slows down so does the musician/rowers.

In electronics, a clock sends a regularly timed pulse to a piece of silicon to tell that silicon how fast to do whatever that piece of silicon does. With an LTC1799 Mod, we hijack that pulse signal and replace it with our own that we can control and change in real time.

As you might guess, different pieces of silicon have different timing of pulses, and different tolerances for how much faster or slower they can run before crashing completely. More about that later, but for now I mention it because the LTC1799 covers a huge range of frequencies; meaning that with a bit of tuning we can use it to hijack just about any oscillator.

Setting It Up

The LTC1799 only comes in an SMD/SMT format. In normal terms: it's incredibly tiny and usually requires special tools to solder (I say usually because some people who are crazy good with a soldering iron could probably do it, but I'm not one of them and I'll assume you aren't either). To make it easier to work with, most folks use a premade board that makes the connections through-hole (or able to be soldered normally). Full disclosure, I make and sell one of these boards (see Figure 1), but you can get versions from other folks or do it yourself if you're comfortable with SMT soldering. This article is written so you could easily do the same thing with any LTC1799 board, regardless of where you get it from.

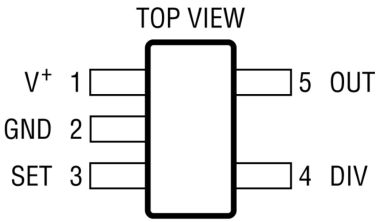


Figure 2: LTC1799 Pinout

First, let's run through what the pins do:

- V+/GND: These pins power the chip. It can take 2.7-5.5 volts, which are super common chip voltages, meaning it's easy to find a spot to hook up power

- SET: Changing the resistance between this pin and V+ is what changes the speed. Values between 10K and 200K Ohms work best.
- OUT: This is where the new clock signal comes out (pretty easy).
- DIV: This one is hard to describe. Basically, DIV means Division. To cover such a wide range of frequencies, this pin can be used to divide the output frequency by 1, 10, or 100. In most cases you'll divide by 1 by connecting the DIV pin to GND. But if your device has a very low frequency oscillator, you may need to leave it disconnected (divide by 10) or connect it to V+ (divide by 100); but in my experience that's fairly rare.

How does it connect to things?

Now that we have an idea of what it does, let's go through how to hook it up in general terms. There will be some specific examples in the last section too.

Power: The V+ and GND connections are pretty easy. Look for points on your device that have a 2.7-5.5v supply (most chips will fall in this range) using a multimeter. Double check that the voltage is only on when the device is turned on to prevent wasting power/draining batteries.

Oscillators: We'll connect the output pin to the existing oscillator. There's three basic ways to connect this, but first a couple tips on finding oscillators:

- They're usually silver metal things shaped like pills or cylinders. Sometimes they also look like blue plasticky rectangles, or a tall silver rectangle with a black circle on top.
- You can also look at how the part is labeled on the circuit board for clues. Usually, an Oscillator will be labeled with an X or a Y (e.g. X1 or Y1).

Figure 3: Some examples of oscillators

Piggyback: The easiest way to connect the LTC1799 is "piggybacking," just connecting the new oscillator to the original one and calling it a day. You can make this connection via a switch to swap between the original and modded clock. This method is what we'll use in the Meowsic example.

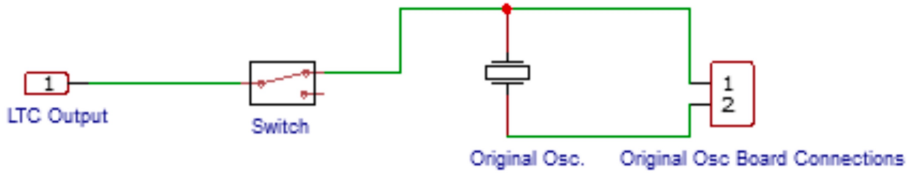


Figure 4: Piggyback Connection Method

One at a Time: Some devices don't respond at all to piggybacking. An electrical engineer could explain why, but I'm not one of those. I usually try piggybacking first, then try this method if that doesn't work or if I feel the results aren't great. It's similar to piggybacking, but we will disconnect one side of the oscillator (you'll have to experiment to find out what side works best) and use our switch to connect either the LTC1799 or the original oscillator. This method is what we'll use in the SK-1 example.

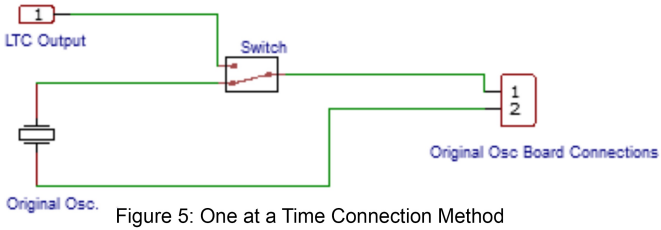


Figure 5: One at a Time Connection Method

Replace: This is the simplest but my least favorite method. You just take out the old oscillator and plug in the LTC1799. I prefer having the option to use the original clock, so I only do this if nothing else works. This method is what we'll use in the REX-50 example.

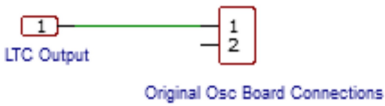


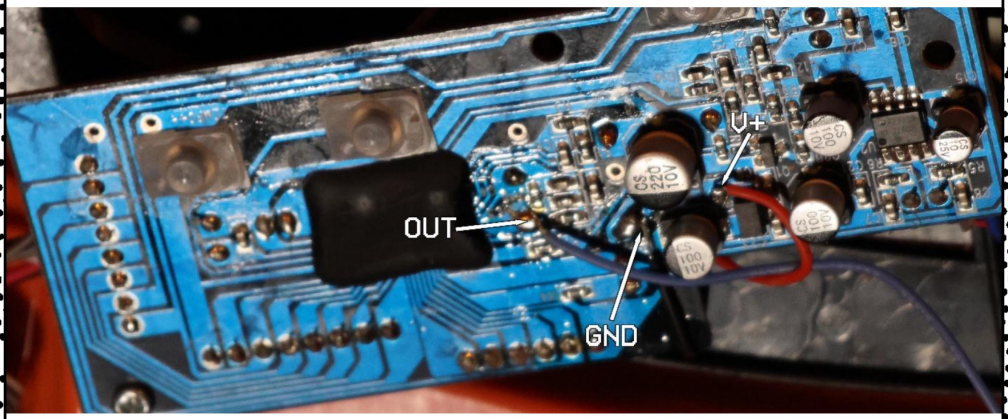
Figure 6: Replace connection method

Resistance: Last, we need to control the resistance between the V+ and Set pins. I think of this in two steps: pot value and trim resistor. The pot value is the value of the potentiometer (pot for short) we'll use to control the clock speed. Something around 100KΩ is usually pretty good, but I like to experiment to find a value that feels right. The trim resistor is just a baseline amount of resistance added to the pot to keep us from turning the speed up too high and crashing the device. The value will vary a lot between projects, so it just comes down to trial and error. I'll give the pot and trim resistor values for each example.

Can you give some examples?

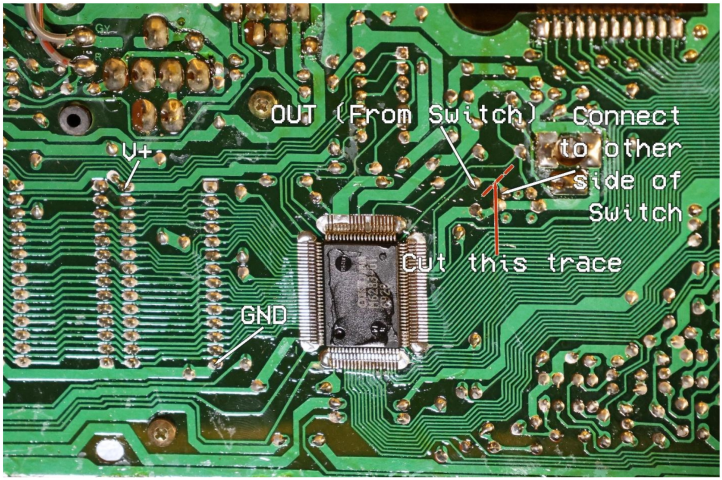
Here's some real world examples of mods to check out. With that you should be ready to go forth and do your own LTC1799 Mods.

Meowsic



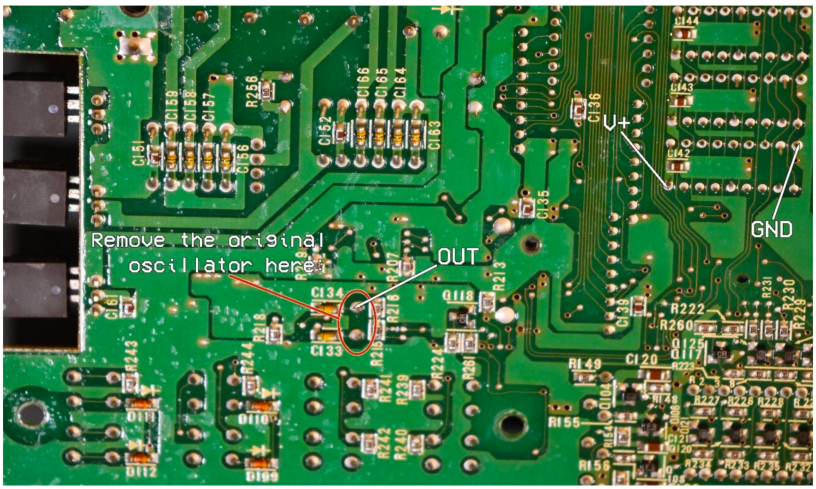
The Meowsic is great because they're fun as heck, easy to find, and cheap! There's a few different versions, but for most of them you'll use a 50K pot and a 5.1K trim resistor. There are a few out there with a teeny silver cylinder oscillator; for these you'll Leave the DIV pin disconnected (instead of connected to GND like all the other examples) and use a 100K pot. I still like a 5.1K trim resistor with those. Regardless of type, we'll use the Piggyback connection method (see Figure 4).

SK-1



The Casio SK-1 is a classic circuit bending project. They used to be cheap, but now are pretty expensive, so bend at your own risk. See the picture below for connection points. This one we connect with the One at a Time method (see Figure 5), using a switch to reconnect the cut trace or swap to the LTC output. I like a pot value of 100K with a 1K trim resistor. You can use the Piggyback method on an SK-1 if you don't want to permanently alter the board, but I prefer the way this method sounds (though this might be a placebo effect, who knows).

Rex-50



The REX-50 is odd because it's the only device where I've had to use the Replace connection method (see Figure 6). But it's pretty simple- just remove the old oscillator and hook up the LTC using a pot value of 500K and a trim resistor of about 8K. It's a fun multi-FX unit on its own, but after modding it can make anything you run through it sound super glitchy!

Thanks for reading!

Hopefully this article is useful on its own, but if you're a video kind of person you can find tons of good stuff on YouTube by myself (@Po8aster) and a ton of other folks as well. I think these mods are a great way to get started making your own experimental gear pretty easily, and hope you'll give it a shot if it sounds interesting to you. Happy bending!

ADDING ARTWORK TO PCBS

Holy Island Audio
holyislandaudio.bigcartel.com

If you're anything like me, you can't resist cracking open a new pedal to peek at what treasures lay hidden inside. Sometimes you're rewarded with more than just a city of electronic components and you get a bit of artwork as well. I'm a firm believer in the positive effect of having good looking gear and as fickle as it may seem, I personally do tend to play gear more if it looks cool!

Since I first started designing my own PCBs I've dreamt about applying artwork to the boards but there was a real lack of resourcing teaching you how to do so online which led to me to learn through trial and error along with some help from fellow builders. Luckily it was a lot simpler than I initially expected so I figured a few people would find it helpful if I explained the process I use.

First things first, you need to be pretty comfortable using Eagle Software. I don't use any other programs for board design so unfortunately I can only help with Eagle- sorry!

When you have settled on your component placements and routed everything to the point where the board is basically ready to send off for fabrication you can add the artwork to the board as a final touch.

I like to use textural artwork that covers the entire board as a reflection of the artwork printed on the outside of the pedal but don't feel like you need to do the same. You could add icons, logos or even just fancy text to your boards to set them apart from other PCBs.

The first thing I do when I've finished laying out my board is to export it from Eagle as an image I can load into Photoshop. When you bring the image into Photoshop it won't be scaled at 100% it's usually a little bigger so you need to measure your board so you can scale down the board to its actual size.

The easiest way to do this is to use the measure tool in Eagle to measure the width and length of your board. You can then create a new document in Photoshop with the same dimensions as your board. To do this click FILE > NEW > Set your width and height and set the resolution to 300 Pixels/Inch (DPI). Copy the board over from your previous document and use the TRANSFORM tool to scale it down to the edges of the document.

I tend to design artwork for the board while in the larger document to retain a bit of clarity through working at a higher resolution.

You can see the spaces on the PCB that are void of components where text or images can lie in the PCB design but don't feel like you have to keep all artwork to those spaces as you can also place artwork over the top of component footprints. If you do put artwork over footprints, Make sure you will be able to remember what component goes where when you're populating the board.
Once you have added your artwork to the Photoshop document you can hide the layer containing the exported image of the PCB.

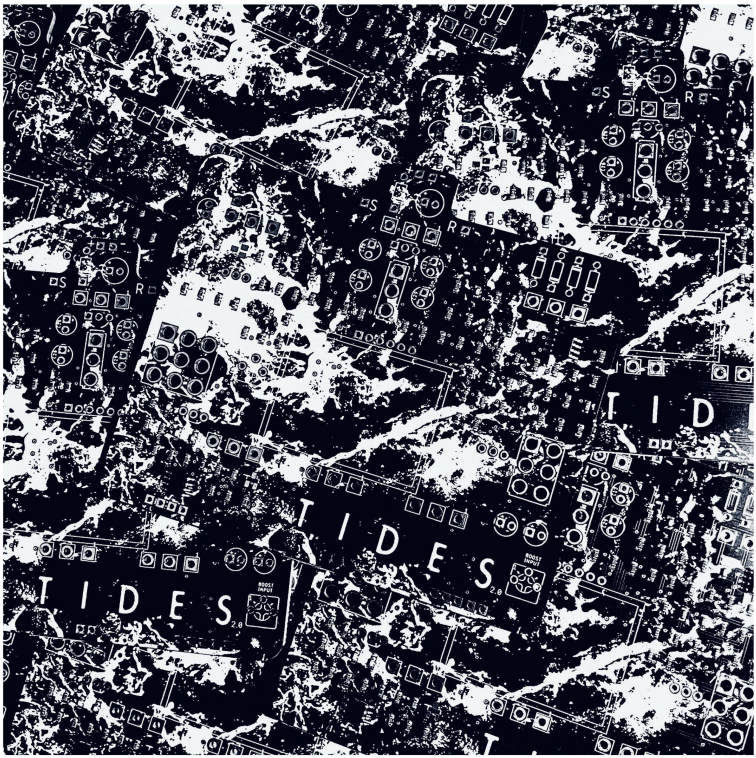
You will now need to convert the document to a bitmap file which can be imported into Eagle. To do this you need to first select MODE from the toolbar and set the image to Grayscale. Once that has converted you need to select MODE from the toolbar again and select Bitmap. You will need to set the Output to 300 pixels/inch and have the Method set to 'Diffusion Dither'. Now you need to save the document as a Bitmap file using the .BMP extension.

Next we need to import that Bitmap we just created into Eagle.

Open the project you want to add the artwork to. Select File from the toolbar then select Import and choose Bitmap. Click OK to the message that pops up and then you will be asked to select the Bitmap file you want to import so choose the artwork you just saved. You will be asked to select what colours you want to use. Select black and press ok. On the next screen you need to set the format to DPI, The unit to INCH, the DPI value to 300 and finally the set the destination layer to 21 and click OK. Eagle will take a minute or two to configure itself and then a small window will appear. Click RUN SCRIPT and Eagle will slowly start importing your artwork. If you are importing a larger artwork this may take a while so be patient while it recreates your artwork line by line.

Once the artwork is fully imported you can move it by choosing RECTANGLE from the selection filter using the GROUP tool to select the entire artwork and then use the MOVE tool to position the artwork where you'd like it.

That's it! Sit back and admire your handiwork and your board is ready for fabrication!



Holy Island Audio's Tides PCBs

Raging Droner

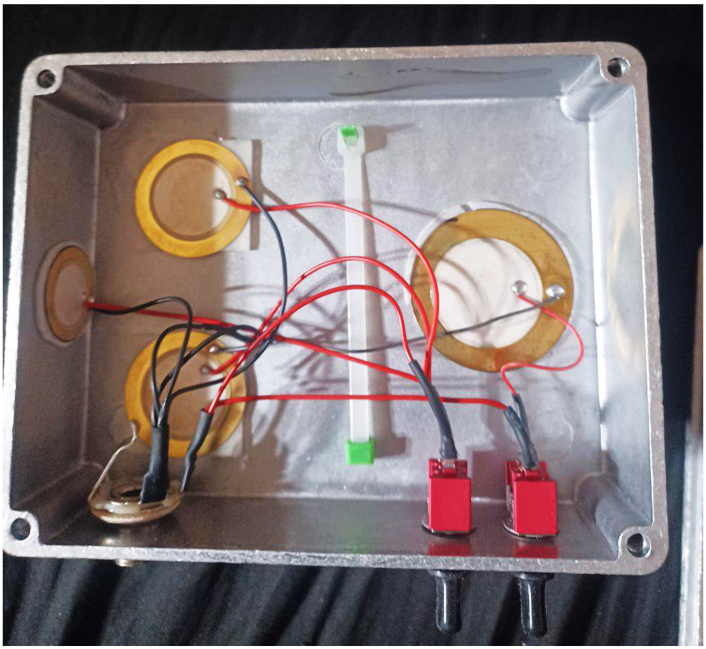
Brett Montez

I became obsessed with noise at a very early age. My older brothers had a shared guitar in the house as far back as I can remember. This would have been the mid 70's. By the early 80's, one of my brothers aspired more toward being a guitarist. He had a lot of pedals. My favorite thing to do, for hours on end, was to run a guitar cable into a distortion pedal and then into a delay pedal. I would touch the cable jack either with my fingers or against something metal and set the delay to repeat endlessly. I would layer sound on sound and marvel at my little noise symphonies. I was hooked!

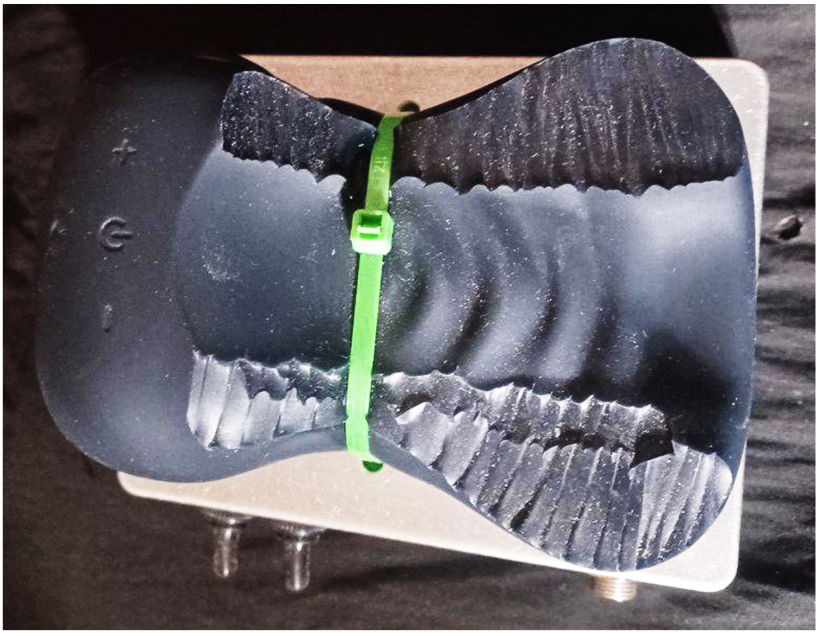
Fast forward to the mid 80's and I acquired a Casio SK1 keyboard/sampler. It was the coolest thing I had ever played with! Add pedals and it became something else, something inspiring. I had a buddy with an old Fostex 4 track and we would make ridiculous recordings. Somehow, I got the notion to crack the sk1 open and "play with its guts". I found out years later that I was doing something called Circuit Bending. It was fascinating! Making that poor keyboard suffer for my "art" became my go-to for so many stupid recordings. As time went on, I became a musician of sorts, singer by trade, but never gave up being experimental. Flash forward again and I ended up scoring some low and mid budget horror films. This would have been the 2000's. Then, it happened.

Somehow, some way, I discovered The Apprehension Engine/Nightmare Machine created by Mark Korvan. This thing was a game changer! I HAD to have one! I had a broken Mackie mixer, gutted it, and used the chassis to add strings and springs and things to make creepy noises. Wired in some contact mics / piezo pickups, and it was awesome! I kept making more and more of these experiment boxes, bought a violin bow and an ebow, and I was in noise heaven!

I got a job in an adult store, and after 10 years, I had acquired many sex toys. Then it clicked, these things make noise! I began experimenting and came up with the device that this article is about. I stripped down a male masturbation toy and loved the way it sounded. Instant drone machine! I mounted it to a guitar pedal enclosure with several contact mics. The results were...stimulating! Multiple speed and pulse patterns ran through pedals and effects units, and the Raging Droner was born! It's a monster!



Raging Droner insides



Raging Droner top

CONTACT

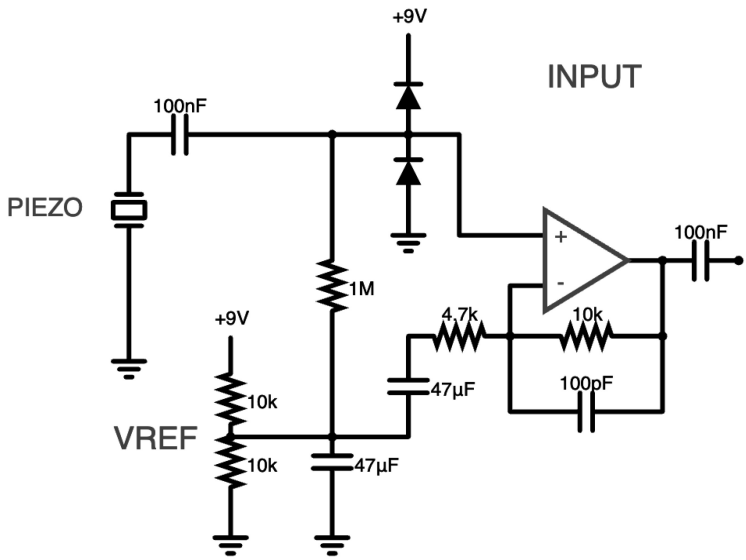
A SIMPLE CHAOTIC MICROPHONE

Noah Sider
nsider1457@gmail.com

A piezoelectric disc, or contact mic, is a small ceramic disc pressed against a thin metal plate, each with a wire attached. A type of electromechanical devices called transducers, piezos convert mechanical energy (being touched) to electrical energy (a very small voltage). Since sound waves are small pressure waves, we can effectively put the disc against a resonating body and hear the output. That is, with contact, we have a microphone! These humble devices need only some gain to find use in everything from cello pickups to harsh noise setups. Let's make a small preamplifier for a piezo disc.

THE PREAMP

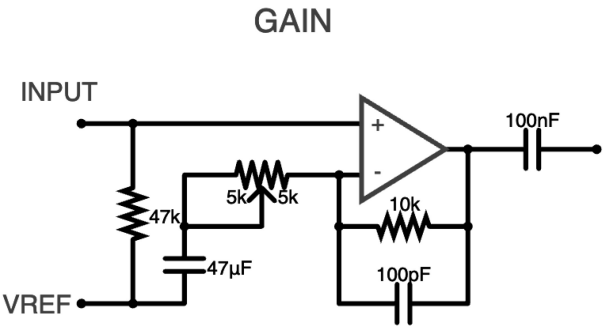
Our first and major problem is that the output of a piezo is very quiet. Since we are simple creatures seeking simple answers, we will simply make this quiet signal louder. Let's do this using a FET input op-amp like a TL072, since they are cheap and plentiful. The following is a basic preamp stage suitable for a piezo:



Our TL072 opamp (as well as a large input resistor) gives us a nice high input impedance which will help us get the most signal possible out of our piezo. Input diodes prevent any dangerous voltage spikes from reaching the opamp. The values shown will give a gain of about 3x. That's just what we need to get the tiny signal from our piezo and pump it up to audible levels.

GAIN STAGE

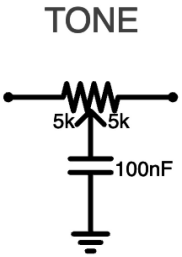
If you just want to listen to the output of the disc quietly this would be sufficient. Let's look for a more interesting sound by adding more gain to distort the output. Here's an additional gain stage:



The gain of this opamp can be changed by varying the potentiometer in its feedback circuit; as it approaches 0, the preamps gain will increase as there is less negative feedback present. The opamp will then try to swing its output closer to the maximum available voltage (the supply voltage!) A 10k potentiometer here will go from unity volume at maximum resistance to extreme distortion and feedback at minimum. If you want to limit the maximum distortion available you may put a small resistor in series with the potentiometer. The maximum available distortion (potentiometer=0Ω) sounds really cool, but it's chaotic and extremely loud. You may wish to not have this sound available. A 470Ω resistor is a good start and you can increase or decrease the value from there.

TONE CONTROL

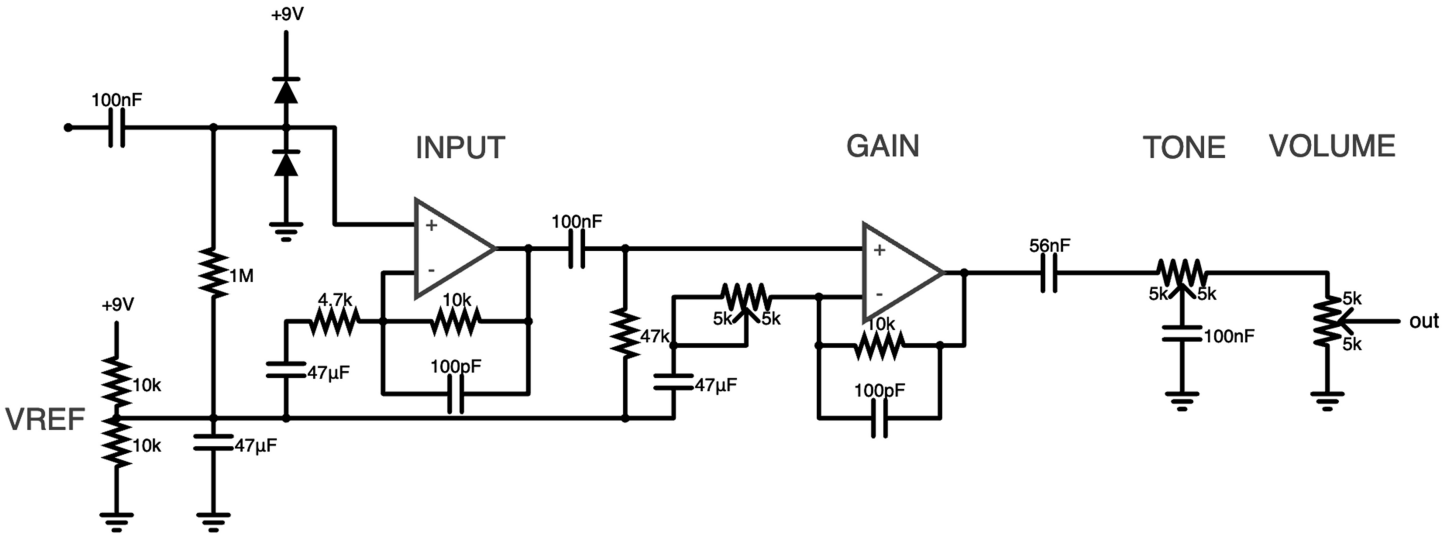
You may also wish to add a simple low pass filter tone control in order to dampen harsh high end frequencies, of which this piezo distortion circuit will have plenty. We can do this with easily with a potentiometer and capacitor to ground:



The value of the pot and capacitor can be changed to set what frequencies are rolled off. Feel free to try whatever values you have lying around!

FINAL RESULT

Let's put our circuit together and admire the result. Here we have a piezo disc, an input stage to bring it up to level, a distortion stage to inspire mayhem and excitement, and a tone control to smooth things out. We'll add a volume control to the end and be finished:



Contact: Final Circuit
Noah Sider

INTERVIEW: PSYCHICEYECLIX

https://psychiceyeclix.wordpress.com/
https://www.etsy.com/shop/Psychiceyeclix

Damian Lintell-Smith A.K.A. Psychiceyeclix is a multi-instrumentalist, sound artist, performer, and circuit bender based in Manchester, UK. In his own words, "I have been making/creating music as Psychiceyeclix since 2001, making 9 ep's/albums released on Cyborg Recordings, Doubledgescissor, Tingo Tongo Tapes, Future Music, Kiff Recording, Junkyard Productions, Fature Music, Dismissive Records, Linear Obsessional Records, Videopunks." As well as releasing a plethora of music, his etsy shop is home to hundreds of circuit bent toys and devices. Everything from voice changing masks to electric kazoos are available for purchase. I was eager to get to know more about his circuit bending background and process. In early January 2025, he agreed to a phone interview for this premier issue.

How did you first get into electronics and circuit bending?
I kind of went to college to study electronics as a teenager and worked in electronics to begin with. And I was the bass player in experimental rock bands. And for the late 90s, I was doing weird stuff like re-wiring a samplers play buttons to foot pedals and stuff like that. So that was kind of circuit bending, but I had never heard of circuit bending at that point. So I'd always try to be doing different things, like make a guitar lead in the shape of a 'Y' so I could play into two amplifiers. I used to do stuff like that, and I continued playing in fucked up electronic rock bands. And I think I got into circuit bending around 2007 or 2008. I think I saw some videos on YouTube of circuit bent Casio SK-1's and stuff like that. So yeah, that's how I got into it. Just looking for a more unique sound, cause that was the music that I was into. I was always into bands that sort of pushed the envelope a little bit.

What kinds of bands were you listening to?
As a teenager I was listening to Primus, The Jesus Lizard, Mr. Bungle, Fugazi. Mr Bungle were one of the biggest influences along with Melt Banana, the Japanese band. I saw those in 1999 with my drummer at the time, and they were like nothing we'd seen before. They were doing something completely unique, so they were a big influence on me years ago.

Do you remember the first toy that you ever circuit bent?
I think it was one of the Mix-Me DJs. It's this toy with a little keyboard on the side, and it had like six trigger buttons, and a little spinning disc in the middle. It's a pretty classic circuit bending thing. So I think that's where I started, yeah.

Are there specific types of devices that you especially enjoy messing with? Or is it whatever you can get your hands on type of thing?
I think older stuff. There's more possibilities with older stuff. Stuff from like the 70's/80's/90's was the golden era where you could still do stuff. You could still modify and reuse components and stuff like that. From the 00's I think that's when they started introducing that black blob. Instead of being normal components, there would just be a black blob with no exposed legs or

FART
SOUND BOX
MEGA DRIVE II
GAMERS
SOUND BOX
Let's Play!
Sound Book
KORG
monotron
Duo
SCARFACE
IN YOUR POCKET
PSYCHICEYECLIX
GNARLYPHONE
DSP Glitchverb
Awesome Mix Vol. 2

PSYCHICEYECLIX
CIRCUIT CIRCLE
8 BIT INVADERS
Sleep Select
Loop
Jog

anything like that. So that limits what you can do to a device. So the older the better, that's my general rule with finding stuff. But you can always do stuff, you know, you can always mess up anything. You can add extra circuits to it and make it sound like a ring modulator, or you can add delay boards or little sampling boards or something like that. So there's always something you can do, it's just with the older stuff there's more possibilities.

When you first open up one of these things and look at the circuit board, how do you gauge what you want to do to it? Do you use your fingers, or some other type of tools?
Yeah yeah, more or less. So the first thing you do is look for the pitch resistor, and that will tell you how far you can go with it. If there's no pitch resistor then you have to try other things, you know.

What other things could be possible if there's no pitch resistor?
If there's no pitch, then you can starve the voltage. So you put a pot between the battery and the input to the board. And that can sometimes have a similar effect to pitch, or that can make it stutter and glitch out and stuff. Or you can build a 555-timer that can trigger, and that can have a pitch control-ish…you can add sequencer chips into it, or ring mod chips, or delay chips, anything like that. It depends how far you want to go.

Did you have any mentors or inspirational figures in the world of circuit bending?
I think Reed Ghazala and Igor Amokian. Reed Ghazala is The Godfather of circuit bending. He's the one that coined the term and stuff. And Igor Amakian, he's always really active, always up to good stuff, just a really positive guy.

Do you have recommendations for websites or books for people who are just starting into it?
I mean, just google search circuit bending and that'll get you on your way. It's more about learning how to do things. You need to break a lot of stuff before you get any results. When I first started, I must've broken like ten Speak & Spells on the way to making one good one. You need to break stuff to get to that point where you know what you can and can't do. Some of my more interesting stuff I did when I was learning. Now I have some sort of process in my mind, and there's things that I would've done fifteen years ago that I won't do now, you know? And I won't find the total random sounds by crossing wires that make no sense. So on my way to learning I found some great glitchy devices that I wouldn't find now because I have more of a concrete process to it. Just a completely beginner's attitude often finds the best results.

Any projects of yours that you'd like to mention?
I just released an album, a collab project with a Swedish artist, Trum Trum. He actually owns a festival in Sweden, and he owns a windmill. And he's converted a windmill into an organ. So when a sail goes around, it pumps the bellows of the organ and plays music. So this building is a musical instrument.

Check out Psychiceyeclix + Trum Trum's album, SWENGLITCH, on streaming and digital media platforms!

DRUGS™
CIRCUIT BENT PS2

https://drugstm.shop
instagram: @drugs_tm

The PlayStation 2 (PS2) is renowned for its groundbreaking advancements in gaming, selling more than any other console. Its unprecedented leap in graphics, sound, gameplay, and massive game library contributed to its iconic status, solidifying its place as a cultural touchstone of the 2000s.

DRUGS™, through their unique circuit bending approach, transforms this familiar household item into a psychedelic art generator that empowers VJs, gamers, and nostalgic hobbyists to explore uncharted territories in creative expression.

Making the Obsolete Contemporary

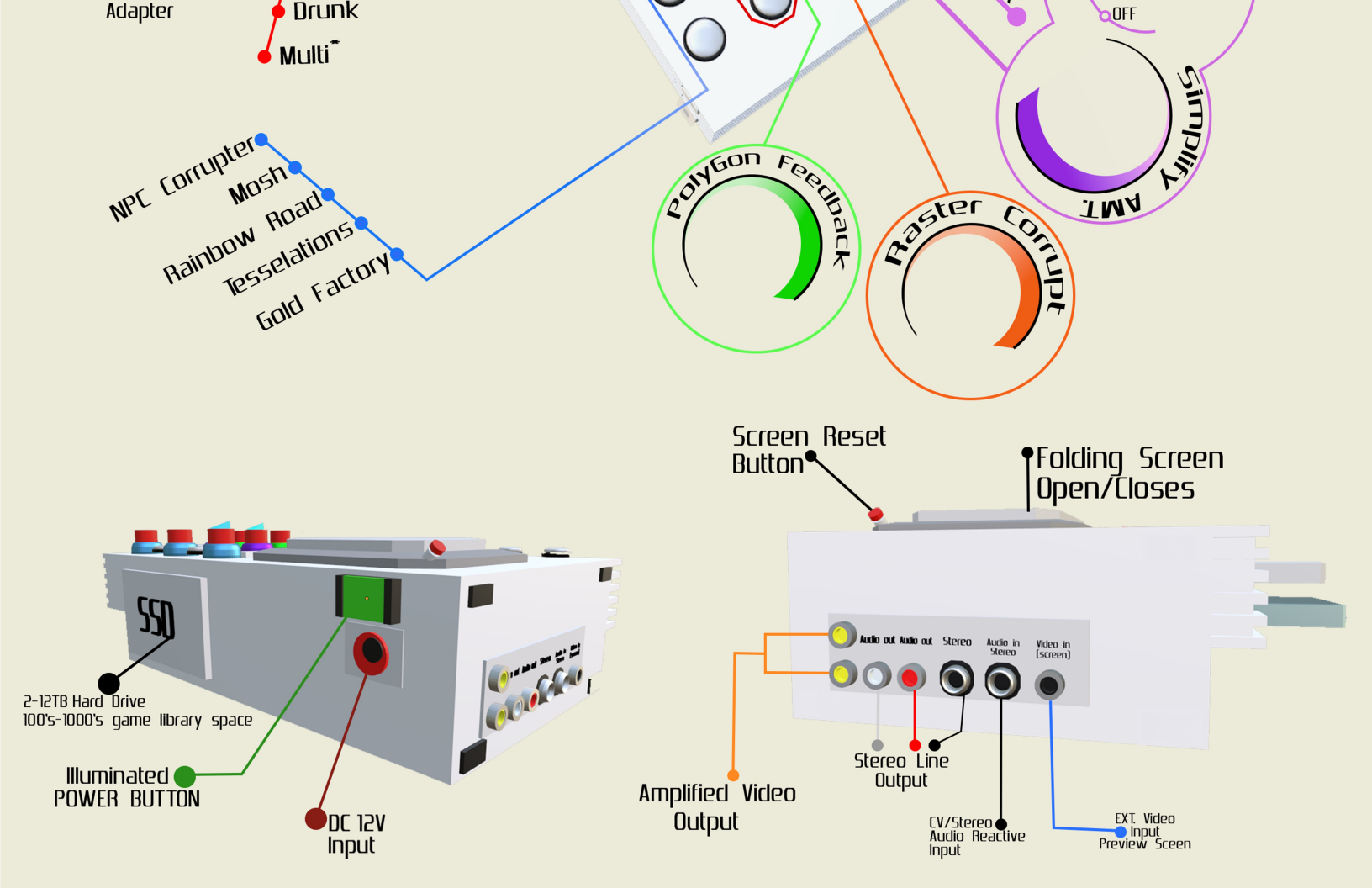
DRUGS™ breathes new life into obsolete PS2 consoles by replacing outdated components like power supplies and disc drives with modern alternatives like DC adapters and SSDs, capable of storing thousands of games. They implement FREE MCBOOT, a custom software developed by Neme, Jimmikaelkael, SP193, and Woon Yung. This software is held in a hacked Memory Card, and in addition to playing ISO game files, DRUGS™ crew member REZIDUAL VIZUALZ has compiled modified versions of applications including a video and music player, a full library of cheat codes unlocking all characters, levels, and programmer hacks to any game you choose. This unlocks the potential for manipulating content well beyond the PS2's original capabilities.

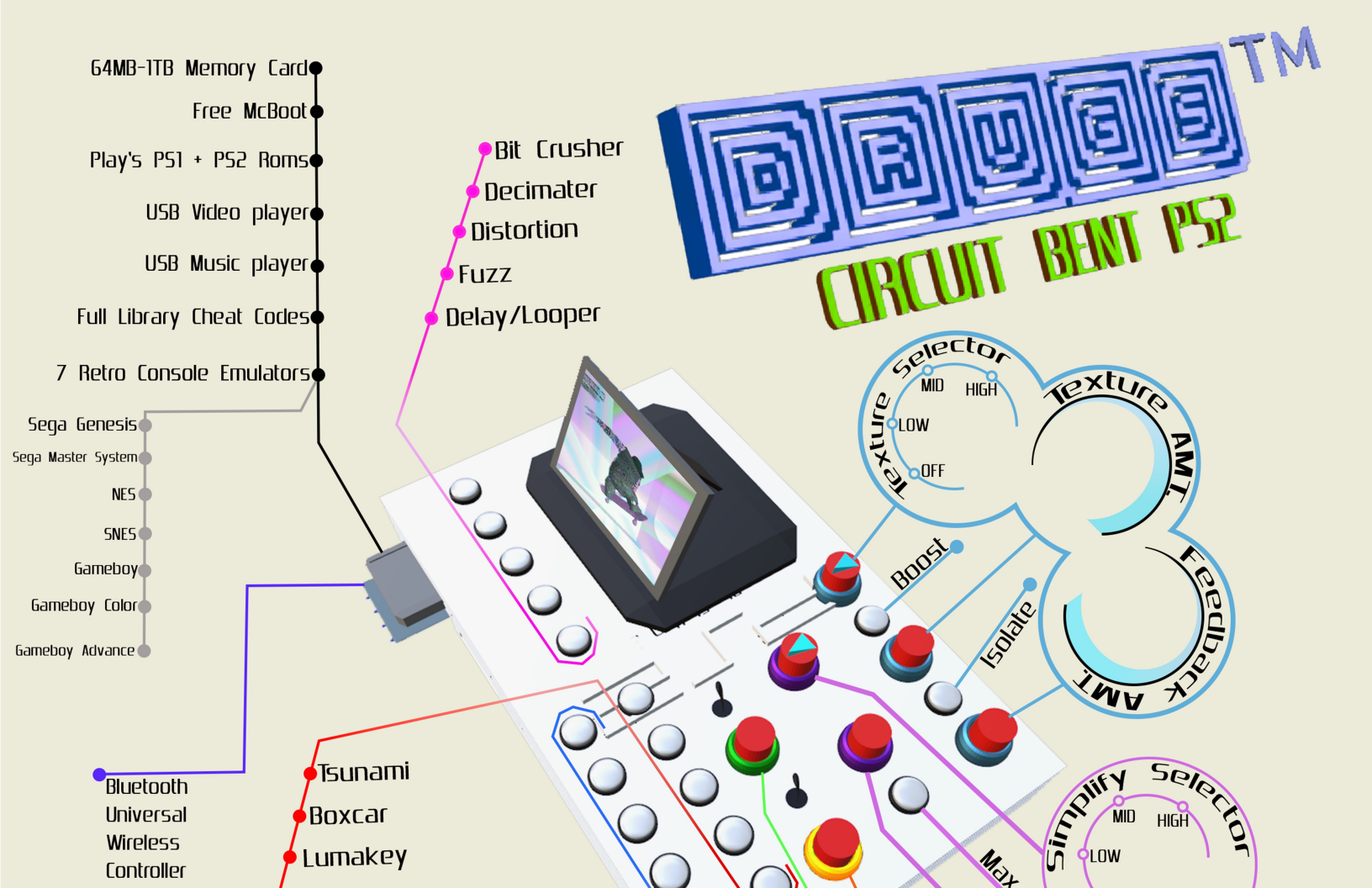
Graphic Glitch Controls

The PS2's graphics are generated from Sony's CPU, the Emotion Engine, sending data packets called "display lists" to the Graphics Synthesizer for rendering. These display lists are created by parallel processing and carry information to render frames, place vertex positions, texture coordinates, and lighting data. The Graphics Synthesizer rapidly renders these lists by rasterizing polygons, applying textures, and calculating pixel colors through hundreds of points of communication between itself and the CPU. After 8 years of research, trial and error, the DRUGS™ leader Colston Byrd has stabilized predictable and intentional control over these data packets. Let's analyze the control layout of their flagship "Speed Runner" model.

Texture Feedback

The top row of controls shown in Fig A. is the texture feedback circuit. This circuit takes the display lists telling the Graphic Synthesizer where to slap specific texture coordinates, and cuts the breaklines of when to stop sending that information. Creating a visual feedback effect. A 4-position rotary switch selects three different destination locations to send this texture feedback information. When the Texture Selection Rotary is in position 1 the circuit is off. When in positions 2,3 and 4 the corresponding knobs in the row light up, signalling that the circuit is on. Each Texture Selection mode varies in intensity, ranging from subtle color shifts to complete visual distortions, enabling creative control over game visuals. The second knob, Texture Amount, controls the amount of texture data sent to each destination. The third knob, Texture Feedback Amount, influences how frequently textures are repeated. Two buttons offer further control: Boost adds an intensity to the Texture Amt knob. The Isolate button allows the Texture Feedback Amount knob to


Adapter
Drunk
Multi**
NPC Corrupter
Mosh
Rainbow Road
Tesselations
Gold Factory
PolyGon Feedback
Raster Corrupt
Simplify AMT
OFF
SSD
2-12TB Hard Drive
100's-1000's game library space
Illuminated
POWER BUTTON
DC 12V
Input
Screen Reset
Button
Folding Screen
Open/Closes
Audio out
Audio out
Stereo
Audio in
Stereo
Video in
(screen)
Amplified Video
Output
Stereo Line
Output
CV/Stereo
Audio Reactive
Input
EXT. Video
Input
Preview Sceen


64MB-1TB Memory Card
Free McBoot
Play's PS1 + PS2 Roms
USB Video player
USB Music player
Full Library Cheat Codes
7 Retro Console Emulators
Sega Genesis
Sega Master System
NES
SNES
Gameboy
Gameboy Color
Gameboy Advance
Bluetooth
Universal
Wireless
Controller
Tsunami
Boxcar
Lumakey
Bit Crusher
Decimater
Distortion
Fuzz
Delay/Looper
DMGS™
CIRCUIT BENT PS2
Texture Selector
MID
HIGH
LOW
OFF
Texture AMT
Feedback AMT
Boost
Isolate
Simplify Selector
MID
HIGH
LOW
Max

be engaged independently even when the rotary switch is off. Each button provides momentary or latching action.

Simplify, aka Chroma Key

Fig A shows the Simplify Circuit, used by VJ's to easily chroma key solid colors into other video sources. This circuit takes display packets targeting where textures go and "starves" them, simplifying visuals. It's ideal for removing unwanted elements from a scene, creating clean aesthetics. A 4-position rotary switch controls the Simplify Circuit: in position 1, it's off; positions 2, 3, and 4 select between 3 Simplify Destinations, each increasing intensity as you move up the knob. When a destination is selected, the circuit turns on, lighting up the corresponding knobs. The second knob controls the effect strength. A "Max" button maximizes the effect for quick cleaning or chroma keying.

PolyGon Feedback

The PolyGon Feedback knob, when engaged, repeats the Graphic Synthesizer's rendered shape information, stacking layers of colors and glitching math across your screen. Turning it up enters a world we call "Painting," where the game seemingly disappears and you now are generating beautiful patterns of interactive color, shapes, and textures. This is the cheat code for PS2-style video feedback art and endless graphic design patterns.

Raster Corrupt

This effect alters the mathematical descriptions of lines and shapes before they are translated into pixels, resulting in diverse outcomes depending on how each game rasterizes polygons. This can range from image feedback, injecting new color palettes, or replacing HUDs with lo-fi pixel art styles. This knob is incredibly versatile for both gaming and “painting”.

Tsunami

This button sends top vertex coordinates for certain polygons skyward, turning game characters into DMT-inspired overlords and if you are near water in the game, it stretches to the sky creating a threatening tsunami effect.

Boxcar

Boxcar warps polygon vertex coordinates vertically, transforming cars into boxes and flat landscapes into mountains. This creates a disorienting visual effect by distorting the polygons on top of the warp itself, and is almost guaranteed to give the player motion sickness.

Lumakey

Luma Key replaces vast textures with lighting data, creating a bright white blooming effect. It's not only mesmerizing but also versatile: VJ's can seamlessly swap the white background for external video feeds through their mixing software's Luma key function.

Drunk

Drunk creates a second, less opaque image overlay, phasing it over your original video to induce a disorienting visual effect akin to high blood alcohol content.

Multi*

There are two Multi effects, but each console gets either one or the other: Double Vision (splitting your screen vertically, doubling the video side-by-side), or LP4 (multiplying videos into a tiled triangle layout resembling Ratatat's LP4 album cover). 100% of games are consistent with Double Vision, vs about 10% of games will engage a alternative distortion of polygon math with the LP4 effect.

NPC Corrupter

This glitch strobes incorrect information into the display packets carrying textures to the non-playable characters in games. This gives a subtle and off-puting effect where you and most of your game environment is untouched, except the characters around you periodically blink with colorful static, giving an eerie vibe hinting that you are indeed playing in the Matrix.

Mosh

While the Mosh effect button doesn't actually datamosh your video, the colors that it introduces into your graphics highly resemble this famous video effect.

Rainbow Road

This effect acts as a gain control on the display packet carrying the color information for textures. This distortion introduces a rainbow of neon colors and patterns into your textures, and is great for introducing psychedelic palettes while leaving the polygon shapes very familiar and playable.

Tessellations

This effect is a repeating version of the Rainbow Road effect, resulting in a stunning technicolour glitch when engaged. Turn both Tessellations and Rainbow Road on for an immersive visual experience while leaving all of your polygons intact.

Gold Factory

DRUGS™'s obsession with the aesthetic of gold is now accessible in every game with a single button press. This Alchemic effect turns ** into gold.

Audio Mods

The PS2's SPU2 is a dual-core 48-channel audio chip with built-in reverb and effects. Although it has an internal ROM circuit, earlier models use a separate ROM chip to hold audio samples before playback. This opens up the world of using traditional point-to-point ROM chip circuit bending techniques. DRUGS™ has created five consistent banger audio effects that bring classic circuit-bent style glitches and clipping flavors to in-game samples and music, or your own music loaded via USB. We recommend glitching your own compositions created using a PS2 DAW game.

Bit crusher

This button reduces bit depth and sample rate of the audio signal. The sonic result is a crunchy lo-fi effect that brings dialogue into Speak n Spell territories and music into retro game console realms.

Decimator

The Decimator effect removes a significant amount of data points from the audio signal. This is known in the music world as "downsampling". While similar to the Bit Crusher effect, the Decimator brings its own flavor of aliasing to the table. Making it a powerful tool in the studio, or bringing a more immersive glitch experience to the gaming side of the console.

Distortion

The harmonic clipping in this effect is as glorious as it is unique. and turns in-game or uploaded USB music files into absolute BANGERS.

Fuzz

Just in case you want more grit, saturation, and compression than the already exaggerated Distortion effect, mash that Fuzz button. Engage both Distortion and Fuzz to get sidechain compression to your kicks and bass. This turns your cheesy PS2 tracks into harmonically rich and mangled Death Grip style masterpieces.

Delay/Looper

This effect feeds back the ROM's output address creating glitchy delays, or loops that differ from game to game. Sometimes even level to level within the games. It's a mess, but with purpose, and nearly impossible to recreate in a daw or effects pedal.

Preview Monitor

By installing an on board distribution amplifier where the old power circuit used to be, this custom add-on powers a top mounted foldable LCD preview monitor. A helpful tool for VJ's and art installations. The monitor's power and video input is hardwired internally in the PS2. Additionally the screen’s video signal can be hijacked by plugging in an external signal via the input jack on the side of the console.

Audio Input Control, CV, and Eyetoy Addons

A ¼" or ⅛" TRS stereo jack on the side of the console lets two channels of incoming amplified audio control the respective left and right half of graphic glitch controls on the console. In addition to this hands free way of syncing up visual glitching with music this jack also allows CV trigger input. DRUGS™ also has a CV box attachment that converts all forms of control voltage into safe ways to modulate via the Audio input jack. The crew also makes a circuit bent camera called the Third Eye Toy that plugs into this jack and the usb, acting as a controller for Eyetoy games as well as interacting with every glitch control on multiple new levels.

A Piece of Their Story

DRUGS™ is a passionate crew of circuit benders, artists, engineers and video game designers who are determined to connect the worlds of gaming and glitch art on a new and fun level. In 2024 their lab and leader were devastated by Hurricane Helene, and now that Colston is healed up from multiple broken ribs and punctured organs, the crew is rebuilding a new lab and will be coming out with more PS2s and new bent game console tech for the rest of 2025 and the foreseeable future.

MULTIBAND SATURATOR

Max For Live Development Journey

GMAudio

GMaudio started as a way for me to explore DSP via MaxMSP back in 2018. It soon grew into the concept that I wanted to build M4L devices that were simple yet flexible, incredibly high sound quality, sensibly priced, with lifetime free updates and a focus on user education.

I started with a small amount of programming & graphic design knowledge, 2 years of teaching audio engineering (specializing in electronic music production) and 16 years of music production experience. I'm now an Ableton Certified Trainer, regularly visit research papers, program nearly everything in genexpr codebox, and can work with most programming languages (especially now with the help of AI).

I wont say it's been easy. Max can sometimes be confusing when things are undocumented (although rare). Trying to define where a bug is, whether it be the patch, Live or Max, can also be confusing. The language and concepts in research papers are often misleading, and I've come to realise that I need a balance of human contact in my life rather than sitting at a computer programming all day.

I do everything myself but have had some fantastic people help along the way. Shout out to Wilson Ryan (ideas and moral support on DSP), Nikita Yermolayevs from Monomono (ideas & moral support on UI & DSP), Ess Mattisson from Fors (MaxForLive inspiration and feedback), Rafael Matos da Silva (UI feedback & ideas), Fabien from Tokyo Dawn Plugins (DSP Inspiration and ideas), Geraint Luff from Signalsmith Audio (DSP Ideas and sanity check), Cycling74 Support, the gen~ channel on Discord and of course Graham Wakefield (the grand-daddy of gen~). GMaudio M4L devices are now a major part of my income source and I'm always looking for ways to improve my work.

Music is my life and has given me more than I could imagine. My goal has always been to share that with others and help them create better music, faster. As a music educator I'm limited to the individuals that I have in front of me. As a music producer I can only ever collaborate with one or two people at a time… but with music software, I can play a small part in the music production process for thousands while I sleep, impacting thousands of songs every year… and I'm incredibly grateful for that.

I develop products that I feel are necessary and are not included with Live Suite, the idea being that I produce music and teach my students to create using nothing but Live & my products. I quite like the synths that come with Live but the audio effects I felt could often be improved and are usually my main area of interest. I feel mixing is often over-complicated by producers, always looking for ways to improve their music rather than actually creating it. so I aim to simplify and refine modern mixing techniques and use my devices to indirectly teach about fundamentals of audio engineering.

The design goal is always:

- It should help, not hinder, music production
- All parameters should have a wide and musical range
- Things of minor importance should be set to an ideal compromise
- Digital artifacts should be at an absolute minimum
- All parameters should support modulation where suitable
- Devices should feel part of the Live ecosystem
- Should be incredibly easy to use, while still including flexibility
- Should perform a single task or concept incredibly well

GMAudio Multiband Saturator

Saturation/Distortion is of great interest to me as I use it on nearly every track in Live. I've been working on various waveshapers for some time and have some unique things in development planned for certain products. Squeeze was a great success and I've gotten so many comments on how much people love it, so I wanted to make something that complimented it. I often do some kind of holiday gift for my users and this time I asked my discord community what device they would like most out of a buchla-style wavefolder, a colourful compressor or mid-side multiband saturator. You can guess which one won.

Refining Anti-aliasing techniques is something I'm constantly working on and ADAA (Anti-derivative Anti-aliasing) is something that I've been playing with for the past couple years. I believe I was the first to implement FIR Polyphase Oversampling in a M4L device (Clipper) and then also with ADAA (HiFreqLimiter - Version 1 & PentaComp 2 - still in beta).

A paper from Martin Vicanek (https://www.vicanek.de/articles/AADistortion.pdf) suggested a slightly different way of looking at ADAA that circumvented some of the pitfalls I was having with typical implementations. Laurent De Soras (aka Firesledge - D16) from the Audio Programmers Discord - DSP channel, helped me understand it better. I was testing MV's implementation and realised that it could work well with 3x Oversampling but the IIR compensation filters didn't meet my design criteria, as I often want to include the ability to listen to the difference (aka delta) and run things in parallel, which would require linear phase and integer sample latency.

I then realised that if I refactored the ADAA implementation to get just the filtering effect (avoiding the non-linearity), I could then derive FIR compensation filters, which in total would give me 3 samples latency. This made it even more ideal for 3x oversampling because it meant that after oversampling, the result would be exactly 1 sample latency, and aligning signals would be a piece of cake. This is now a kind of standard for anti-aliasing for me moving forward, it has caveats depending on design requirements, but it has incredibly good antialiasing properties and is a good compromise in CPU efficiency.

For multiband processing, things must be very efficient. You wouldn't want to run 5x oversampling on every band, so I thought it would work perfectly in a multiband saturator. Because it is linear-phase it makes multiband processing easier to manage, allows things to run in parallel, the delta signal is exacting and it could be used in mid/side. Combining that with the crossovers from Squeeze and viola, the Multiband Saturator prototype was born.

Two things I often use in my music are a 6dB IIR highpass filter in the side channel to help mitigate mono compatibility issues (cheaper than FIR and doesn't rotate the phase destructively) and a spectrum analyser monitoring the sum and difference (mid/side) so I can easily see where there might be opportunities for poor phase coherence in stereo. Both of these were always going to be a part of this Multiband Saturator. I also wanted to keep things as simple and as fast as possible while utilizing minimal device chain width, which informed my decisions with the UI. One pet peeve of mine is saturation plugins/devices that are not level compensated, so you can't just apply a subtle amount of saturation in a mixing context and hear what it sounds like without messing about with (often unintuitive) gain controls.

Through testing and refinement, I arrived at the final product, which much like Squeeze, has a surprising number of use cases. For instance, in a mastering context you could use it to softclip bands very subtly to increase loudness, while also checking/improving phase coherence (if sides exceed the mid), increase the width of certain bands if there is room to do so, make subtle adjustments for balance and do it all in parallel as to not overdo the processing… all within a single device with a very unassuming interface.

I already have proof of concept about how I can take things further in terms of DSP, but I need to nut out the details about how this will work with the UI and the CPU overhead they will cost. Like how about saturating only the transients or sustain of a certain band/channel, or having a more unique character parameter (perhaps more akin to analog) for each band, rather than just basic asymmetry?

Multiband Saturator was meant to be a thank you gift for completing a survey, but instead I just opened it up to everyone for a limited time. The survey is coming to help define the direction of GMaudio and Multiband Saturator is now a paid product.

The Future

I have more prototypes and ideas than time at this point, but there are updates on the way for VSEQ, Ducker, PentaComp, TransD, Universe, Noizyer & Fusion. Version 2 of both Squeeze & HiFreqLimiter were released in January 2025. There are also three new products in development, a single band Saturator with unique waveshapers and useful colouration controls that aren't often found in a typical saturator, a multiband brickwall limiter that is low latency but sounds tight and punchy with minimal distortion, and an analog style EQ with some unique filters and saturation options.

I want to include my users in what I do as much as possible, because I cherish sharing knowledge and having social contact. Fostering a community where more people can take part in helping others would be an ideal scenario that aligns with my mission, while also allowing me to accomplish more of the things I'm best at while keeping education at the forefront.

I've regularly toyed with the idea of developing VST/AU plugins. Max For Live is great and people really like the device chain integration, portability and ease of sharing devices when collaborating, but sometimes dealing with 2 two interdependent frameworks (Live & Max) has its issues. I adore the speed of prototyping things in genexpr and will always use it to explore DSP but I can't help but wonder if there are more ideal ways to realise the final product.

One thing I've always wanted to do is create an iOS AUv3 plugin that can be used as end of chain mastering and metering for DJ's, broadcast and Live performance. I have also toyed with the Electro-Smith Daisy platform and developing eurorack modules, but supply chains got a little hairy during Covid and I rarely use my Eurorack system at the moment, so I'm unsure of that pursuit at this stage.

What I do know is that I am somewhat addicted to exploring DSP and the problem solving aspects of product development, and I don't think I can stop. I also really enjoy communicating my ideas in various ways, and quite enjoy things like writing. AI as a collaboration partner is a recent realization for me and it's helping me iterate faster than ever before. It's buying me more time to learn new skills and focus more time on the things I want to do.

Curiosity is what led me here and will continue to carry me forward. I wish the same for you and I hope you find abundance in your curiosities. Happy music making and circuit breaking, punks!

The End[1]

Thanks for reading! You can get 50% off Multiband Saturator here:
https://robertkgm.gumroad.com/l/gma-mbs/circuit-punk_50
If you'd like to see behind the scenes of what I do,
or have a query about my products, then join my Discord here:
https://discord.gg/fWWyNcNtWh
If you just want to get in touch, you can email me :
robertk@fixationstudios.com.au

[1] https://fixationstudios.com.au/circuit-punk-gmaudio/

ANALYZING AND RECREATING

REVERSE BIAS TRANSISTORS

Sean Porio
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I'm Sean, an audio electronics designer at Death By Audio. We design all kinds of effects at our company but our bread and butter is gnarly, twisted, and over-the-top fuzzes and distortions. In the course of some recent work on a fuzz circuit, I came across an interesting problem and a somewhat birdbrained and dastardly solution. I'd like to explain the process of identifying and solving some of the mysteries of the reverse-active transistor in hopes it might inspire you, reader, to try applying some funky properties of transistors to make some very crazy sounding pedals.

To prevent taking up too much paper with an electronics crash course, this entry is going to assume a relatively solid familiarity of transistors, diodes, and general electronics to understand, but I would invite anyone who might know even the foggiest smidge about resistors and capacitors to try to follow along. I've been in your position many times and I promise that after reading this and jumping to other resources (I strongly recommend Douglas Self, Moritz Klein, Electrosmash.com, and Vocademy Electronics), upon reading it again you'll reveal to yourself another small layer and reading and learning more will pull back another each time. Tear off the layers at your own pace. Let's jump in.

The Common Emitter Amplifier (AKA The Simplest Fuzz You Can Make)

Fig. 1 shows a somewhat simplified version of a BJT transistor circuit known as a Common Emitter Amplifier. This is the basis of countless fuzz circuits across the ages and it's where our story begins. The transistor-savvy among you might object that this circuit is very prone to distortion, has unpredictable gain and operating point with different transistors in the same batch, and has a somewhat high output impedance. You would be correct. We are in fuzz land and not hi-fi-ville, but please keep your electronics sensibilities about you because they will definitely come in handy.

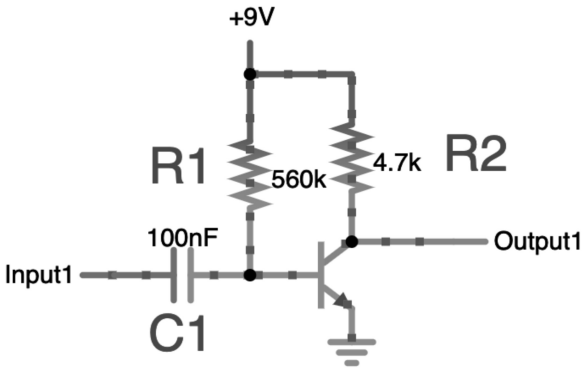

Fig. 1: A simple common emitter amplifier

This circuit operates by sending a small amount of signal current into the base of the transistor through R1. This causes a large amount of current to be pulled into the collector from the power supply through R2. The sum of those two currents eventually all passes down to

ground through the emitter. The proportion of current pulled into the collector I_C versus the current sent into the base I_B is called the β, h_{FE}, or "gain" of the transistor. This typically ranges from around 100-300 in modern silicon transistors. In other words, if 1 μA of current is sent into the base, somewhere between 100-300 μA will be pulled into the collector. Using this setup, we can send a small amount of DC current through R1 to turn the transistor on, then send a small audio signal into the circuit through C1. The current into base will change as the audio waveform hits its peaks and troughs causing a much larger change in current through the collector. When the collector current increases, a large voltage drop will be created across R2 while a decrease in collector current will cause a smaller voltage drop. This means that as the voltage at the base increases, the voltage drop across R2 will increase greatly, causing the voltage at the collector to swing down from the supply rail. And thus, you now have an inverting amplifier! This circuit is fundamental to countless fuzz and distortion circuits going as far back as the Rangemaster, Fuzz Face, and the Tone Bender. "Nicer," "fancier" applications of this circuit are used in almost every op-amp, audio IC, and amplifier circuit you can buy.

A few months ago, I was looking to make some revisions to an old Death By Audio pedal that used this circuit. This pedal and a few others often had an issue that different batches of transistors would make the pedal sound completely different. This would lead to folks on the team scrambling to find which batches worked with which pedals and modifications being made to new circuit board orders to make sure the resistors were tuned to the latest batch. As we were about to order another batch of circuit boards, I wanted to see if I could get to the bottom of it. I looked at the pedal, looked at the board, double- and triple-checked the schematic, and I saw something I did not expect. Something that explained why we needed entire batches of transistors tested, thrown aside, and sourced from the far ends of the Earth to get right: The transistors were upside down.

What the Hell is Reverse Active Mode?

"Reverse-Active" is a term you sometimes see floating around on forums and scantly addressed in technical documents. After all, why discuss how to use a transistor the wrong way? It refers to the behavior of a transistor when used "upside-down;" using the collector as the emitter and vice-versa. Why would you want to do that? I do not have a great answer to be perfectly honest, but I can at least promote that it's captivating to use something like a transistor the wrong way when you're trying to make a gnarly-sounding effect. In the pedal I was working on and many other Death By Audio fuzzes, it tends to sound quite nice. Whether or not you agree, I can try to explain the method and theory behind the madness.

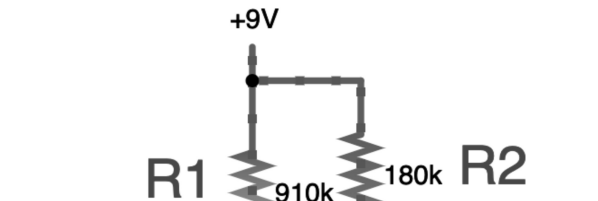

Fig. 2: A common-emitter amplifier with a reverse-active transistor. The genesis and the original sin of Death By Audio fuzz.

For all intents and purposes, we can still use a BJT transistor “upside-down” and it will *basically* do the same thing. That is, you can connect your input signal to the base of an NPN as usual, but connect the *collector* to ground and the *emitter* to the load resistor. Depending on the transistor you’re using, you will probably get currents flowing that somewhat resemble a transistor’s behavior when used “correctly.” The tradeoff, and the reason you’re *supposed* to use it the other way, is that the β of the transistor tends to be *very* low—in the range of single digits, as opposed to the 100s you might get using it the right way around. This is a pretty measly amount of gain, but if you want to use a transistor in a very broken way, we can still apply the common-emitter configuration with some values adjusted to get an amplifier. With a similar base current causing a much smaller collector current, we will need a much larger R2 in order to get the same amount of voltage gain. This drastically increases the output impedance of the circuit, making the amplifier less than stellar at making things louder depending on the application. But an amplifier, it absolutely is!

For all its shortcomings, it had been discovered at the company long ago that the particular arrangement of certain transistors with the resistor values shown in Fig. 2 tends to sound very good, especially when several are chained together in series[1]. Whether or not it’s a “good idea” to use reverse-active amplifiers with their issues is open to interpretation but we are dedicated to bringing wild-sounding circuits in all of our products and this happened to be one method that has stuck around and been applied in many designs. We also definitely would not want to be changing the sound of any pedal going back decades in our catalog to do something “the right way.” If it sounds right, it is right.

The problem is, it didn’t always sound right.

When Upside-Down Gets Ugly

The serious issue about using the transistors this way is that the reverse-active gain of transistors are often not controlled for in any way. While many other specifications of the transistor might be tuned to be at least somewhat consistent, reverse-active gain is almost never specified on the datasheet and I would seriously doubt that it’s ever even measured. As a result, every batch may have a wildly different gain and there is no way as a consumer to know what it will be. After all, as a semiconductor manufacturer, why the Hell would you try to control the reverse β? What lunatics would design something using it?[2] This revealed why entire batches of transistors needed to be tossed when building these pedals. Different batches, even from the same manufacturer, would have massive differences in this reverse β, many having only a gain of 1, making them completely useless to us. That in combination with the somewhat “rugged” (highly β-dependent) design of this simple amplifier block makes for wild and unacceptable differences when the β of a batch of transistors is different.

After much time over the years finagling with replacing resistors on boards, changing designs to better fit new batches of transistors, it seemed now that I better understood the source of the problem, we could potentially find a solution to make the pedals consistent. The use of “upside-down” transistors in our designs was something I had discovered early in my time with the company and I had originally written it off as an unusual quirk that was going to be a beast for us to spar every once in a while whenever we needed to restock on transistors. I had a lot of other things to do at the time[3] and so I moved on with the precarious nature of those circuits just scratching the back of my mind every few months or so. I did attempt to create more transistor amplifiers that had the same overall voltage gain but I could not precisely

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[1] Countless Death By Audio schematics ranging in quality from dubious and shockingly accurate are available all over the internet. I’ve used them myself for quick checks when our copies aren’t handy and I don’t feel like getting up from my desk.

[2] Us!

[3] This included solving an issue where a certain pedal occasionally emitted magic smoke when being tested

replicate the distortion characteristics of the original pedal. Suitable fuzzes, maybe, but never the spitting image of this specific sound. A reliable, very low-β transistor is really what we needed to continue making these pedals without affecting their sound and it didn't seem like there was any way to get one. It was while working on some other non-fuzz designs that I became more familiar with another transistor circuit that ended up being the key to putting this hair-pulling to rest.

Enter The Current Mirror

The current mirror is another essential electronics building block, but in guitar pedals, you're not likely to see it used discretely in the signal path[4]. I did not use it in a guitar pedal design until it became useful getting LM358s to better output 0V for some CV stuff. After playing around with the circuit and finding other applications for it though, I got the idea to see if I could make something useful in fuzzland out of it. Out of that fiddling around came a near-perfect replication of a reverse-active transistor without using anything upside down, backwards, or inside out.

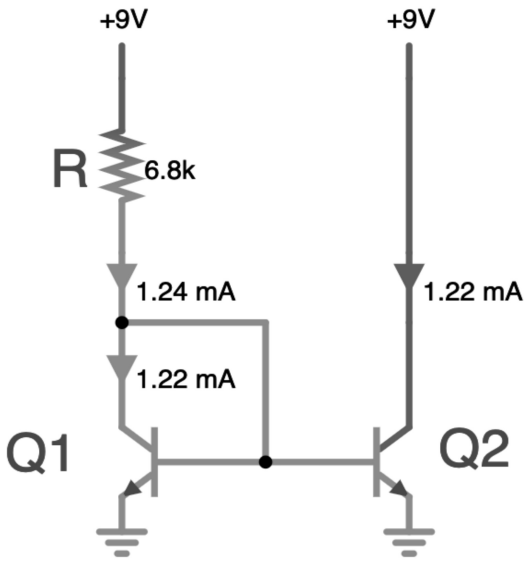


Fig. 3: A simple current mirror circuit

A simple current mirror functions based on the property that transistors generally have a very predictable relationship between the voltage drop from their base to emitter (V_{BE}) and their collector current (I_C). This property is used to "copy" a current from one transistor to another by connecting their bases and emitters. Fig. 3 shows an example circuit. Ignoring the effects of Q2 for a moment, consider the left side: Q1 and R. When the collector and base of a BJT transistor are connected, its base-emitter junction behaves like a regular diode. When a resistor is connected to 9V and then to a diode to ground, we should expect some current to flow through and see voltage drops across the resistor and diode. That's exactly what we see here with Q1; 1.2 mA flows through R and down through the base-emitter junction to ground,

[4] I qualify that sentence with "discretely" because they are very often seen in the construction of operational amplifiers (op-amps), but you're unlikely to see the inner workings of an op-amp used in a guitar pedal schematic unless it was created by a maniac. If you'd like to become a maniac like me, start reading op-amp datasheets in your spare time.

causing a voltage drop across the resistor. Here's the magical thing though: the V_{BE} that's created here when 1.2 mA passes through Q1 will be the same V_{BE} that will *induce* 1.2 mA to be pulled into the collector of another transistor of the same type. By connecting Q1's base to Q2's base and connecting both emitters to ground, we've made the V_{BE} of both transistors the same and therefore their collector currents will be identical. As Fig. 3 shows though, there is a small 20 μA difference in the total current pulled through R and the current pulled into the collector of Q2. This is the base current of Q1 and Q2. Having very high β here would minimize this differential and actually improve the performance of this circuit (who'da thunk?). Nothing in life or electronics is free.

Okay, big whoop. How is this helpful in recreating a low-beta transistor? Well, if we call the node where the bases meet "*the* base," the node where the emitters meet "*the* emitter," and the collector of Q1 "*the* collector," we have basically just made a transistor with a gain of one. Still seeming pretty useless, but hold onto that abstraction. The thing is you can keep adding on transistors with their bases and emitters connected and you'll keep getting the same collector current, making more and more "copies" of the reference current. Fig. 4 shows how this can get us our sacred, shitty transistor with a β of three. Q1 sets the collector currents of three transistors and if we connect their collectors together, they will pull *triple* the current sent into Q1. Send current into "the base," get triple that current into "the collector." That's a gain of 3! Alakazam.[5]

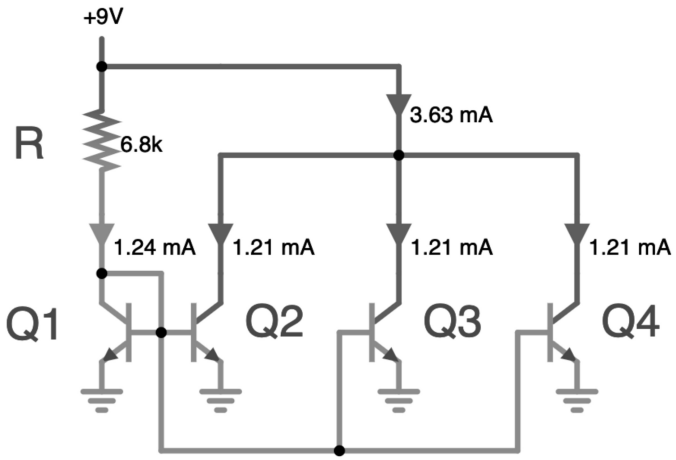


Fig. 4: A current mirror with three transistors tapped onto the same reference; a "current tripler."

With this newfound power, we can finally recreate the reverse-active transistor circuit. When the idea came to me to attempt this, I threw together the circuit shown at the bottom of Fig. 5 to compare to the original amplifier shown above it. After checking to make sure the DC operating points all looked good, I plugged it into my amplifier and guitar to take a listen and I couldn't believe it. Not only did it work, but it sounded almost identical. After a few tweaks, this led me to make a full prototype recreating the entire schematic of that original pedal while replacing all upside-down transistors with the current mirror circuits. And lo, it sounded *indistinguishable*. Even better than perfect though, it sounded that way *every single time*.

[5] A slight caveat is that the precise IV characteristics of this new meta-transistor are not exactly the same as a single transistor of the same type, but the reverse-active amplifiers were fairly inconsistent even when working and I would also note we're making a fuzz pedal so I don't think it's really a concern. It looks like a duck on the oscilloscope and it quacks like one in my headphones.

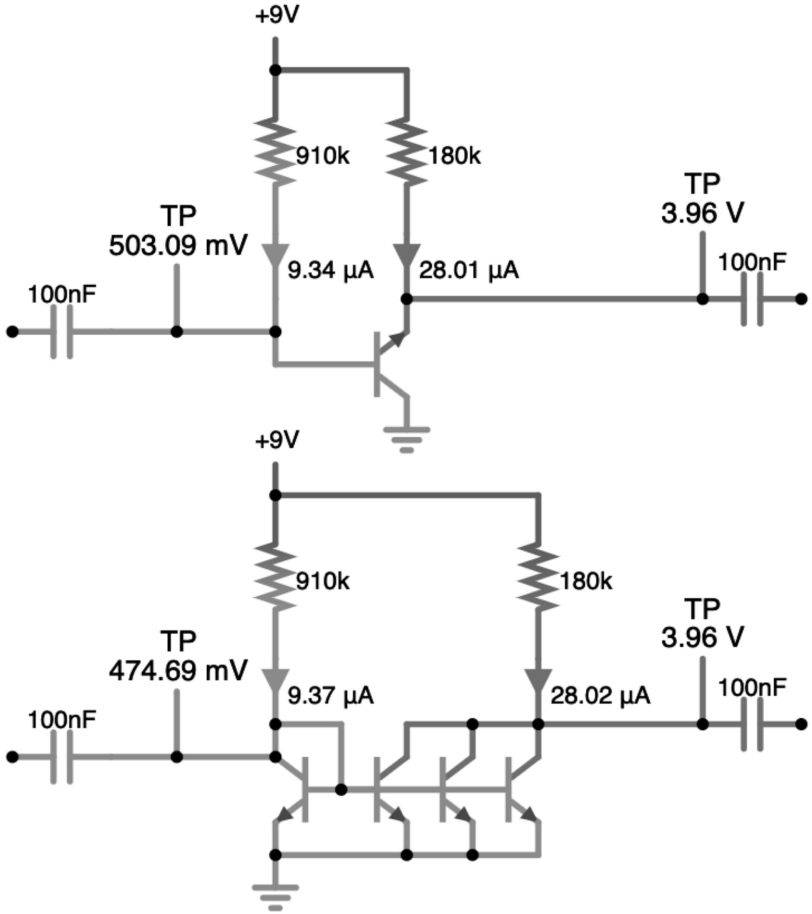


Fig. 5: A reverse-active common-emitter amplifier and its recreation using a current mirror-based current tripler.

Conclusion

What's the lesson of all of this fuss? I don't necessarily think that I'm trying to tell any of you to start using transistors upside-down or try to make a fuzz pedal out of 100 transistors wired as current mirrors. I'm not trying to tell you not to do that either. What I'd really like to say here is that electronics is cool and applying your skills can be incredibly rewarding. I hope you learned something, first about how transistor amplifiers work if you were unfamiliar, but also how to take the building blocks of electronics and bend them to your needs. This is one small example of taking the raw materials of simple circuits to make something that's novel and unique. There are so many fantastic resources available on how to make certain types of audio circuits, but you will absolutely run into something you need that it seems no one has ever made before until you peel it apart to find the ingredients you need to start foraging. Any one of you that's interested in electronics and loves learning new things will find your own ways applying your intuitions and knowledge to solve problems, make music, and create circuits that are wholly original. I would never argue that I'm some kind of electronics mastermind; I am, truly, a jackass. But a jackass that has studied a lot of schematics and datasheets and pulled from them a few nuggets of insight that helped me solve a very wacky problem, and I think any of you reading this can do the same. We're all sitting at benches turning tiny specs of metal and rocks into machines that make music. And that's cool as hell.

GETTING LOOPY WITH TAPE LOOPS

Will Smillie
Moonbeam Music

The advent of digital recording has democratized the process of making records and releasing music. However, it can sometimes leave tracks feeling sterile or lackluster. Cassette tape loops can add titillating textures, wonderful warbles, and ooey-gooey lo-fi beauty to your tracks.

Making a tape loop is easy, and while it may take a couple of tries to get it right, the character is what makes it fun! You can start making your own loops by breaking out an old cassette tape (or a new one). At Moonbeam Music, we recommend high-bias Type II tape for low-noise, high dynamic range recordings.

What you'll need:

- A ruler
- A screwdriver
- A pair of scissors
- A cassette (with screws)
- Some adhesive tape (scotch tape or packing tape)
- A guitar pick (optional)
- A little bit of patience

Steps:

1. Remove the screws from the cassette and gently separate the two halves of the shell, setting the top aside.

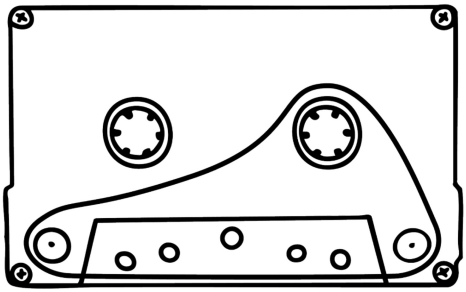
2. Remove the two spools of tape from the cassette shell.

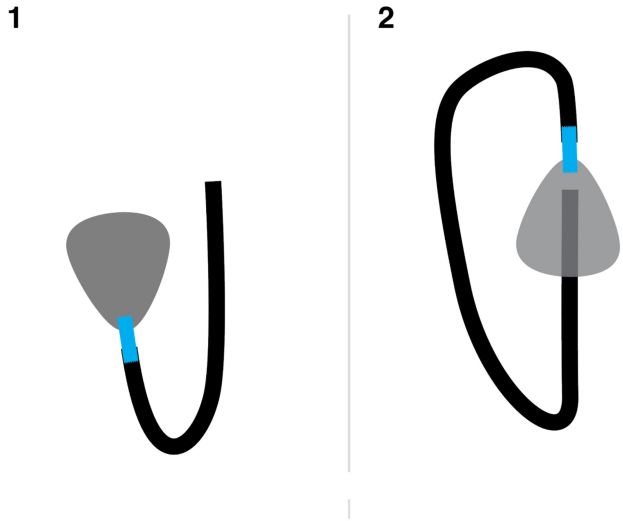
3. Remove the magnetic tape from around the plastic reels/spools, placing the empty reels back in the shell. You can discard about 95% of the tape but keep some to work with.

4. Cut a length of magnetic tape measuring 8 5/8 inches.

5. Cut a piece of adhesive tape about 1/2 inch in length and no wider than the magnetic tape.

6. Apply adhesive tape to one end of the magnetic tape, using a guitar pick (or your fingers) to maneuver and align the ends of the magnetic tape. The adhesive tape should be applied to the inside of the magnetic tape, following its natural curl (the same way it was originally rolled inside the cassette).





7. Form a loop by holding the other end of the magnetic tape in place. Use the guitar pick to align the ends and secure them with the adhesive tape.

8. Thread your new tape loop inside the cassette shell, placing it around the right-most reel and under the guide rollers (at the bottom corners of the cassette). The tape should follow its original path across the pressure pad and magnetic shield.

9. Reassemble the cassette by screwing the top half of the shell back onto the bottom half. If static electricity causes the loop to move out of place, wipe the shell with a dryer sheet to remove any static buildup.

That's it! Your tape loop is ready to roll! You may want to place a small piece of painter’s tape over the erase head inside your tape deck to eliminate playback gaps caused by the physical space between the erase and record heads. If your tape deck won’t engage recording, cover the safety tabs on top of the cassette shell.

Happy looping! Don’t forget to embrace the charm and character of your creation—wabi-sabi at its finest.

3PG

Revolutionizing the Guitar: The 3PG Story

https://3dprintguitars.com
instagram: @3dprintguitars

In the world of musical instruments, innovation often takes a backseat to tradition. But 3D Print Guitars (3PG) is rewriting the rules, crafting guitars that are built to last a lifetime. By seamlessly blending advanced thermoplastics with space-grade metals, 3PG is giving the guitar a modern overhaul—one that prioritizes durability, playability, and sustainability. This isn't just a guitar; it's a statement about the future of instrument design.

The Heart of the Guitar: 3D-Printed Bodies

At the core of every 3PG guitar lies its revolutionary 3D-printed body. Each body is crafted from recycled plastic filaments, making sustainability a cornerstone of the design. What truly sets these bodies apart, however, is the precision and speed of their creation. Printed in one seamless piece using custom-built 3D printers—down to the firmware—these guitars are the result of years of research and programming.

The proprietary tool paths developed by 3PG enable unparalleled layer adhesion, resulting in stunning, holographic finishes that are as durable as they are beautiful. Additive manufacturing also allows for the creation of custom semi-hollow

internal structures, significantly reducing weight and material waste without compromising strength.

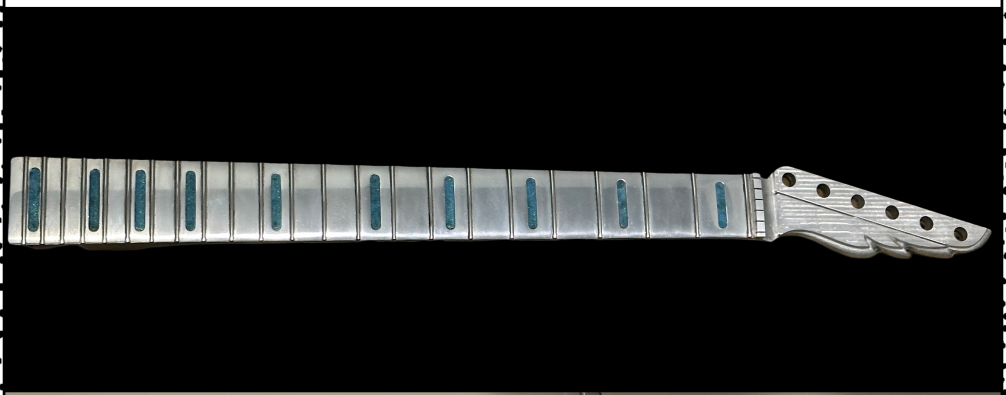
But even the most advanced materials have their challenges. During rigorous testing, 3PG identified a common issue with 3D-printed bodies: the warping nature of plastic under extreme conditions. To combat this, the team designed a custom aluminum frame that extends from the neck pocket to the bridge. This frame is bolted into the body, eliminating warping and ensuring a flawless playing setup that stands the test of time.

Currently, 3PG offers bodies in two thermoplastics: PETG (Polyethylene Terephthalate Glycol) and PC-CF (Polycarbonate Carbon Fiber). PETG provides an endless array of colors, from opaque to transparent holographic, while boasting high temperature and impact resistance. The absence of paint makes these bodies scratch-resistant, ensuring they look as good as new even after years of use. For those looking for a truly indestructible body 3PG offers PC-CF with even higher impact and temperature resistance! Capable of withstanding 250F and blunt impacts from concrete.

The Neck: A Masterpiece of Precision

A revolutionary body deserves an equally groundbreaking neck. Traditional wooden necks, while classic, are prone to moisture absorption, warping, and cracking. They often require truss rods, which need regular adjustments to maintain playability. 3PG asked a simple question: What if the neck was made from a material so rigid that a truss rod became unnecessary? The answer: guitar-playing heaven.

Enter the "Hemera," 3PG's first aluminum neck model. Milled to a precision of one-thousandth of an inch, the Hemera features a hollowed-out internal chamber for maximum weight reduction. It is paired with an aluminum fretboard secured by 27 bolts and adorned with stainless steel frets. Unlike traditional nickel frets, stainless steel frets are virtually indestructible, ensuring they never wear down. The result is a neck that delivers brilliant tone, unparalleled sustain, and a playing experience that redefines what a guitar can be.



The Vision Behind 3PG

3D Print Guitars is the brainchild of Eric “Early” Gima, a visionary engineer with a passion for innovation. Early’s journey began in the mountains of Asheville, North Carolina, where he worked for Aluminati Guitar Company. There, he pioneered the technology behind 3D-printed guitar bodies. However, driven by a desire to make sustainable, high-performance guitars more accessible, Early broke away to found 3PG.

Today, all 3PG guitars are produced off-grid in a solar-powered facility in Corozal, Puerto Rico. Every machine is custom-built for maximum efficiency, ensuring that the production process is as sustainable as the guitars themselves.

A Guitar for the Future

3D Print Guitars isn’t just creating instruments; it’s crafting a legacy. By combining cutting-edge materials, innovative design, and a commitment to sustainability, 3PG is redefining what a guitar can be. Whether you’re a professional musician or a passionate hobbyist, a 3PG guitar offers more than just superior playability—it offers a glimpse into the future of music.

In a world where tradition often overshadows innovation, 3PG stands as a testament to what’s possible when creativity meets technology. This is the guitar, evolved.

GENERAL GUITAR SPECS

- 6.5-8.5LB Total Weight Depending on Body Type, Filament Density and Hardware
- Chambered 6061 Aluminum Neck w/ Hemera Winged Headstock
- Precision Milled Neck Relief
- State of the Art 3d Printed Fully Sealed Internally Chambered Semi Hollow Body
- PETG Colored, or Carbon Fiber PolyCarbonate Body
- Polished Aluminum Fretboard
- Jumbo Stainless Steel Frets
- High Quality Locking Tuners
- Bolt on Aluminum Nut
- Zero Fret Design for Open Chord Stability, Lowest Possible Action, Easy String Gauge Swaps
- Custom Pickups or Customer Preference.
- 3-Way Toggle
- Chrome or Black Bridge Standard Bridge with Saddles
- 25.5 Scale with 23 Frets
- 12” Radius
- D Shaped
- 500K Dunlop Super Pot Lifetimes
- HardCase Shell
- Starting at 1500$ USD

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Write for us!

Circuit Punk is written by people like you!
Got a weird instrument, DIY effect/synth,
or other music tech project?
Write 500-2000 words about it and send it to:
submissions@circuitpunk.org

MARCH 2025
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